The Journal of the Acoustical Society of America

Vol. 138, No. 3, Pt. 2 of 2, September 2015 www.acousticalsociety.org

170th Meeting
Acoustical Society of America

Hyatt Regency Jacksonville Riverfront Hotel
Jacksonville, Florida
2–6 November 2015

Table of Contents on p. A5

Published by the Acoustical Society of America through AIP Publishing LLC


Come See Us at the ASA Exhibit
Booth #5
www.pcb.com/acoustics
YOU GET MORE
THAN JUST WHAT'S
INSIDE
OUR MICROPHONES
Prepolarized
Free-field Microphone
Model 378B02
IEC 61094-4 compliant test and measurement microphone for audible range applications
PERFORMANCE
NOT JUST A MICROPHONE
Inside every microphone is 40 years of sensor design and manufacturing expertise. Exceeding all expectations, our price, support, warranty and delivery are unmatched.
UNBEATABLE PRICE
24/7 TECH SUPPORT
BEST WARRANTY
FAST SHIPMENT
PCB PIEZOTRONICS INC.
Toll Free in USA 800-828-8840
E-mail info@pcb.com
Website www.pcb.com
New Product Finder
Shop Online Now!
www.pcb.com/ShopSensors

THE JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA
Postmaster: If undeliverable, send notice on Form 3579 to:
ACOUSTICAL SOCIETY OF AMERICA
1305 Walt Whitman Road, Suite 300,
Melville, NY 11747-4300

ISSN: 0001-4966
CODEN: JASMAN

Periodicals Postage Paid at
Huntington Station, NY and
Additional Mailing Offices

THE JOURNAL OF THE
ACOUSTICAL SOCIETY OF AMERICA

170th Meeting: Acoustical Society of America

138, No. 3, Pt. 2, 1727–1990, September 2015

CODEN: JASMAN ISSN: 0001-4966

INFORMATION REGARDING THE JOURNAL

Publication of the *Journal* is jointly financed by the dues of members of the Society, by contributions from Sustaining Members, by nonmember subscriptions, and by publication charges contributed by the authors' institutions. A peer-reviewed archival journal, its actual overall value includes extensive voluntary commitments of time by the *Journal* 's Associate Editors and reviewers. The *Journal* has been published continuously since 1929 and is a principal means by which the Acoustical Society seeks to fulfill its stated mission—to increase and diffuse the knowledge of acoustics and to promote its practical applications.

Submission of Manuscripts: Detailed instructions are given in the latest version of the “Information for Contributors” document, which is printed in the January and July issues of the *Journal*; the most current version can be found online at scitation.aip.org/content/asa/journal/jasa/info/authors. This document gives explicit instructions regarding the content of the transmittal letter and specifies completed forms that must accompany each submission. All research articles and letters to the editor should be submitted electronically via an online process at the site <http://jasa.peerx-press.org/>. The uploaded files should include the complete manuscript and the figures. Authors are requested to consult the online listings of JASA Associate Editors and to identify which Associate Editor should handle their manuscript; the decision regarding the acceptability of a manuscript will ordinarily be made by that Associate Editor. The *Journal* also has special Associate Editors who deal with applied acoustics, education in acoustics, computational acoustics, and mathematical acoustics. Authors may suggest one of these Associate Editors, if doing so is consistent with the content or emphasis of their paper. Review and tutorial articles are ordinarily invited; submission of unsolicited review articles or tutorial articles (other than those which can be construed as papers on education in acoustics) without prior discussion with the Editor-in-Chief is discouraged. Authors are also encouraged to discuss contemplated submissions with appropriate members of the Editorial Board before submission. Submission of papers is open to everyone, and one need not be a member of the Society to submit a paper.

JASA Express Letters: The *Journal* includes a special section which has a separate submission process than that for the rest of the *Journal.* Details concerning the nature of this section and information for contributors can be found at the online site http://scitation.aip.org/content/asa/journal/jasael/info/authors.

Publication Charge: To support the cost of wide dissemination of acoustical information through publication of journal pages and production of a database of articles, the author's institution is requested to pay a page charge of $80 per page (with a one-page minimum). Acceptance of a paper for publication is based on its technical merit and not on the acceptance of the page charge. The page charge (if accepted) entitles the author to 100 free reprints. For Errata the minimum page charge is $10, with no free reprints. Although regular page charges commonly accepted by authors' institutions are not mandatory for articles that are 12 or fewer pages, payment of the page charges for articles exceeding 12 pages is mandatory. Payment of the publication fee for *JASA Express Letters* is also mandatory.

Selection of Articles for Publication: All submitted articles are peer reviewed. Responsibility for selection of articles for publication rests with the Associate Editors and with the Editor-in-Chief. Selection is ordinarily based on the following factors: adherence to the stylistic requirements of the *Journal*, clarity and eloquence of exposition, originality of the contribution, demonstrated understanding of previously published literature pertaining to the subject matter, appropriate discussion of the relationships of the reported research to other current research or applications, appropriateness of the subject matter to the *Journal*, correctness of the content of the article, completeness of the reporting of results, the reproducibility of the results, and the significance of the contribution. The *Journal* reserves the right to refuse publication of any submitted article without giving extensively documented reasons. Associate Editors and reviewers are volunteers and, while prompt and rapid processing of submitted manuscripts is of high priority to the Editorial Board and the Society, there is no a priori guarantee that such will be the case for every submission.

Supplemental Material: Authors may submit material that is part supplemental to a paper. Deposits must be in electronic media, and can include text, figures, movies, computer programs, etc. Retrieval instructions are footnoted in the related published paper. Direct requests to the JASA office at jasa@aip.org for additional information, see http://publishing.aip.org/authors.

Role of AIP Publishing: AIP Publishing LLC has been under contract with the Acoustical Society of America (ASA) continuously since 1933 to provide administrative and editorial services. The providing of these services is independent of the fact that the ASA is one of the member societies of AIP Publishing. Services provided in relation to the *Journal* include production editing, copyediting, composition of the monthly issues of the *Journal*, and the administration of all financial tasks associated with the *Journal*. AIP Publishing's administrative services include the billing and collection of nonmember subscriptions, the billing and collection of page charges, and the administration of copyright-related services. In carrying out these services, AIP Publishing acts in accordance with guidelines established by the ASA. All further processing of manuscripts, once they have been selected by the Associate Editors for publication, is handled by AIP Publishing. In the event that a manuscript, in spite of the prior review process, still does not adhere to the stylistic requirements of the *Journal*, AIP Publishing may notify the authors that processing will be delayed until a suitably revised manuscript is transmitted via the appropriate Associate Editor. If it appears that the nature of the manuscript is such that processing and eventual printing of a manuscript may result in excessive costs, AIP Publishing is authorized to directly bill the authors. Publication of papers is ordinarily delayed until all such charges have been paid.

CODEN: JASMAN ISSN: 0001-4966

The Journal of the Acoustical Society of America

CONTENTS

ACOUSTICAL SOCIETY OF AMERICA

The Acoustical Society of America was founded in 1929 to increase and diffuse the knowledge of acoustics and promote its practical applications. Any person or corporation interested in acoustics is eligible for membership in this Society. Further information concerning membership, together with application forms, may be obtained by addressing Elaine Moran, ASA Office Manager, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, T: 516-576-2360, F: 631-923-2875; E-mail: elaine@acousticalsociety.org; Web: http://acousticalsociety.org

Subscription Prices, 2015

	U.S.A. & Poss.	Outside the U.S.A.
ASA Members	(on membership)	
Institutions (print + online)	$2225.00	$2390.00
Institutions (online only)	$2000.00	$2000.00

The Journal of the Acoustical Society of America (ISSN: 0001-4966) is published monthly by the Acoustical Society of America through the AIP Publishing LLC. POSTMASTER: Send address changes to *The Journal of the Acoustical Society of America,* 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Periodicals postage paid at Huntington Station, NY 11746 and additional mailing offices.

Editions: *The Journal of the Acoustical Society of America* is published simultaneously in print and online. Journal articles are available online from Volume 1 (1929) to the present at http://asadl.org.

Back Numbers: All back issues of the *Journal* are available online. Some, but not all, print issues are also available. Prices will be supplied upon request to Elaine Moran, ASA Office Manager, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; FAX: (631) 923-2875; E-mail: elaine@acousticalsociety.org.

Subscription, renewals, and address changes should be addressed to AIP Publishing LLC - FMS, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Allow at least six weeks advance notice. For address changes please send both old and new addresses and, if possible, your ASA account number.

Claims, Single Copy Replacement and Back Volumes: Missing issue requests will be honored only if received within six months of publication date (nine months for Australia and Asia). Single copies of a journal may be ordered and back volumes are available. Members—contact AIP Publishing Member Services at (516) 576-2288; (800) 344-6901, membership@aip.org. Nonmember subscribers—contact AIP Publishing Subscriber Services at (516) 576-2270; (800) 344-6902; E-mail: subs@aip.org.

Page Charge and Reprint Billing: Contact: AIP Publishing Publication Page Charge and Reprints—CFD, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; (516) 576-2234; (800) 344-6909; E-mail: prc@aip.org.

Document Delivery: Copies of journal articles can be purchased for immediate download at www.asadl.org.

TECHNICAL PROGRAM SUMMARY
*Indicates Special Session

Monday morning
*1aAB Comparative Neurophysiology of the Auditory System I: Session in Honor of Albert Feng
*1aAO Acoustics of High Latitude Oceans I
1aNS Environmental and Community Noise
*1aPA Phononic Metamaterials I
*1aSP Direction of Arrival (DOA) Estimation, Source Localization, Classification, and Tracking Using Small Aperture Arrays I

Monday afternoon
*1pAA Sustainability and Acoustics
*1pAB Comparative Neurophysiology of the Auditory System II: Session in Honor of Albert Feng
*1pAO Acoustics of High Latitude Oceans II
1pBA Medical Ultrasound and Imaging
*1pNS Soundscape and Its Application
*1pPA Phononic Metamaterials II
*1pSP Direction of Arrival (DOA) Estimation, Source Localization, Classification, and Tracking Using Small Aperture Arrays II

Monday evening
*1eID Tutorial Lecture on Sonic Booms: A “Super” Sonic Saga

Tuesday morning
*2aAA Acoustics of Multifamily Dwellings
2aAB Bioacoustics Across Disciplines: Detecting and Analyzing Sounds
*2aAO Passive-Acoustic Inversion Using Sources of Opportunity I
*2aBA Wave Propagation in Complex Media: From Theory to Applications I
*2aEAa Vector Sensors: Theory and Applications
2aEAb Analysis of Sound Sources
*2aED Effective and Engaging Teaching Methods in Acoustics
*2aNS Damage Risk Criteria for Noise Exposure I
*2aSA Flow-Induced Vibration
2aSCa Speech Production Potpourri (Poster Session)
2aSCb Analysis and Processing of Speech Signals (Poster Session)
2aSP Detection, Feature Recognition, and Communication

Tuesday afternoon
*2pAAa Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio I
*2pAAb Measuring Sound Fields in Healthcare Environments
2pABa Bioacoustics Across Disciplines: Emitting Sound
2pABb Bioacoustics (Poster Session)
*2pAO Passive-Acoustic Inversion Using Sources of Opportunity II
*2pBA Wave Propagation in Complex Media: From Theory to Applications II
2pEA Analysis of Sound Sources, Receivers, and Attenuators
*2pED Take 5’s
*2pID Guidance From the Experts: Applying for Grants and Fellowships
*2pNS Damage Risk Criteria for Noise Exposure II
*2pPA Acoustic Characterization of Critical Phenomena
*2pSA Structural Acoustics and Vibration in Buildings
2pSCa Forensic Acoustics, Speaker Identification, and Voice
2pSCb Speech Perception Potpourri (Poster Session)

Wednesday morning
*3aAAa Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio II
*3aAAb Worship Space Acoustics: Three Decades of Design
*3aAO Munk Award Lecture
*3aBAa Sonothrombolysis
3aBAb Therapeutic Ultrasound, Microbubbles, and Bioeffects I
*3aEA Test Facilities and Acoustic Calibration
*3aED Undergraduate Research Exposition (Poster Session)
*3aNS Role of Fit-Testing Systems for Hearing Protection Devices
3aPA General Topics In Physical Acoustics I
3aPP Psychological and Physiological Acoustics Potpourri (Poster Session)
*3aSA Nonlinear Techniques for Nondestructive Evaluation
3aSC Various Topics in Speech Communication
*3aSP Random Matrix Theory in Acoustics and Signal Processing
*3aUW 50 Years of Underwater Acoustics under ASA

Wednesday afternoon
*3pAA AIA CEU Course Presenters Training Session
3pBA Therapeutic Ultrasound, Microbubbles, and Bioeffects II
*3pED Acoustics Education Prize Lecture
*3pID Hot Topics in Acoustics
3pSC Speech Style and Sociophonetics (Poster Session)

Thursday morning
*4aAA Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance 1
*4aAB Avian Bioacoustics
*4aBA Numerical and Analytical Modeling of Medical Ultrasound I
*4aEA Acoustic Material Characterization Methods
*4aMU Stick-Slip Processes in Musical Instruments
*4aNS Thoughts on the Next Generation of ANSI Loudness Standards
*4aPA Launch Vehicle Acoustics I: Acoustics of Launch Vehicles and Supersonic Jets
*4aSC Development of Speech Production and Perception Across the Lifespan
*4aUW Environmental Variability Impact on Shallow Water Acoustics I

Thursday afternoon
*4pAAa Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance II
4pAAb Architectural Acoustics Potpourri
*4pAB Bioacoustics Research In Latin America
*4pBA Numerical and Analytical Modeling of Medical Ultrasound II
4pEA Layered Media
*4pMU Acoustical Evolution of Musical Instruments
*4pNS Wind Turbine Noise
*4pPA Launch Vehicle Acoustics II: Analysis and Modeling of Noise from Supersonic Jets
4pPP Pitch, Loudness, and Other Perceptual Phenomena
*4pSA Novel Treatments in Vibration Damping
*4pSCa Advancing Methods for Analyzing Dialect Variation
4pSCb Contributions to the Special Sessions (Poster Session)
4pSP Algorithm, Analysis, and Beamforming
*4pUW Environmental Variability Impact on Shallow Water Acoustics II

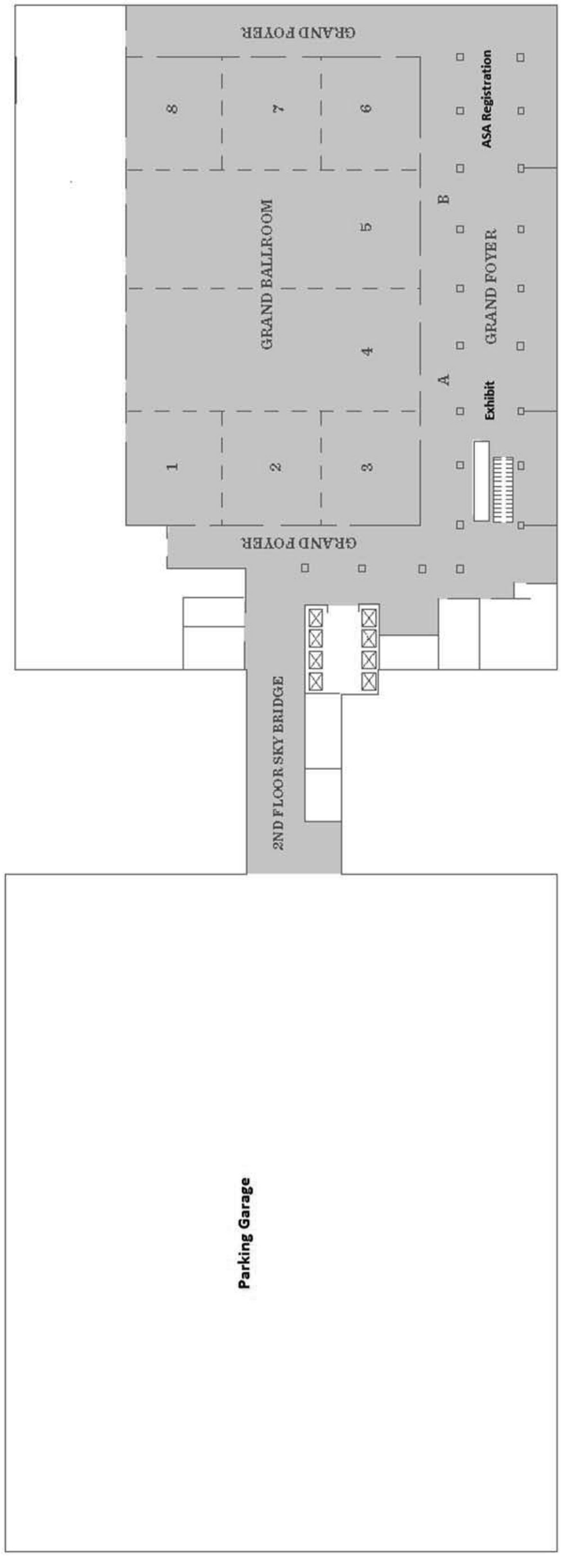
Friday morning
*5aAB New Discoveries in Bat Vocal Communication
5aAO Topics in Acoustical Oceanography
5aMU General Topics in Musical Acoustics
5aNS Noise Potpourri
5aPA General Topics in Physical Acoustics II
5aSA General Topics in Structural Acoustics and Vibration
5aSCa Intonation, Tone, and Prosody (Poster Session)
5aSCb Foreign Accent and Multilingual Speech Production and Perception (Poster Session)
5aUW Communications, Transducers, Target Response, and Nonlinear Acoustics

SCHEDULE OF STARTING TIMES FOR TECHNICAL SESSIONS AND TECHNICAL COMMITTEE (TC) MEETINGS

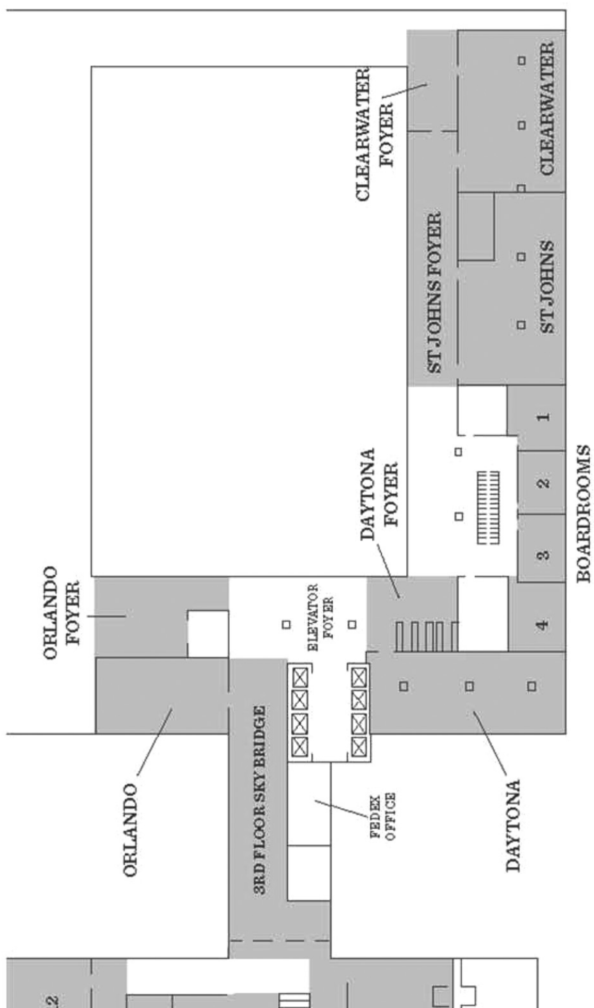
	M am	M pm	M eve	Tu am	Tu pm	Tu eve	W am	W pm	W eve	Th am	Th pm	Th eve	Fri am
City Terrace 7	1aSP 8:30	1pSP 1:30		2aSP 9:00			3aSP 9:00		TCSP 8:00		4pSP 1:30		5aAO 9:00
City Terrace 9	1aAB 9:00	1pAB 1:30		2aAB 9:00	2pABa 1:00	TCAB 7:30				4aAB 8:30	4pAB 1:00		5aAB 8:20
Clearwater		1pBA 1:30		2aBA 8:00	2pPA 1:10		3aBAa 8:00 3aBAb 11:15	3pBA 1:00	TCBA 7:30	4aBA 8:30	4pBA 1:30		5aPA 8:00
Daytona				2aSA 8:35	2pED 1:30 2pID 3:15	TCSA 7:30	3aSA 8:00				4pSA 2:45		
Grand Ballroom 1	1aNS 8:30	1pNS 1:00		2aNS 8:30	2pNS 1:00		3aNS 8:30			4aNS 8:30	4pNS 1:15	TCMU 7:30	5aNS 9:00
Grand Ballroom 2					2pAAb 2:55		3aAAb 9:35	3pID 1:00		4aMU 9:00	4pMU 1:00	TCNS 7:30	5aMU 8:30
Grand Ballroom 3		1pAA 2:00		2aAA 8:00	2pAAa 1:20	TCAA 7:30	3aAAa 8:00	3pAA 1:00		4aAA 8:55	4pAAa 1:00 4pAAb 3:20		
Grand Ballroom 6			1eID 7:00	2aED 8:00	2pSCa 1:30		3aSC 9:00	3pED 2:10		4aSC 8:50	4pSCa 2:00	TCSC 7:30	
Grand Ballroom 7					2pSA 1:00	TCPP 7:30					4pPP 1:30		
Grand Ballroom 8				2aSCa 8:30 2aSCb 10:30	2pABb 3:30 2pSCb 3:30		3aPP 8:00	3pSC 1:45			4pSCb 4:05		5aSCa 8:30 5aSCb 10:30
Grand Ballroom Foyer							3aED 10:00						
River Terrace 2	1aAO 8:30	1pAO 1:15		2aAO 8:15	2pAO 1:30	TCAO 7:30	3aUW 7:45 3aAO 10:55			4aUW 8:00	4pUW 1:00	TCUW 7:30	5aUW 8:00
Orlando				2aEAa 8:00 2aEAb 10:15	2pEA 1:30	TCEA 4:30	3aEA 8:00			4aEA 8:30	4pEA 1:00		5aSA 8:30
St. Johns	1aPA 8:35	1pPA 1:00			2pBA 1:00	TCPA 7:30	3aPA 8:00			4aPA 8:30	4pPA 1:35		

Hyatt Regency Jacksonville

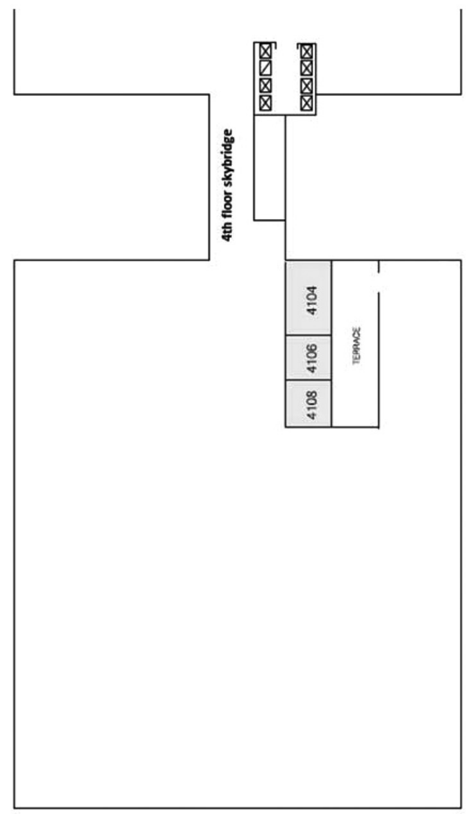
Second Floor

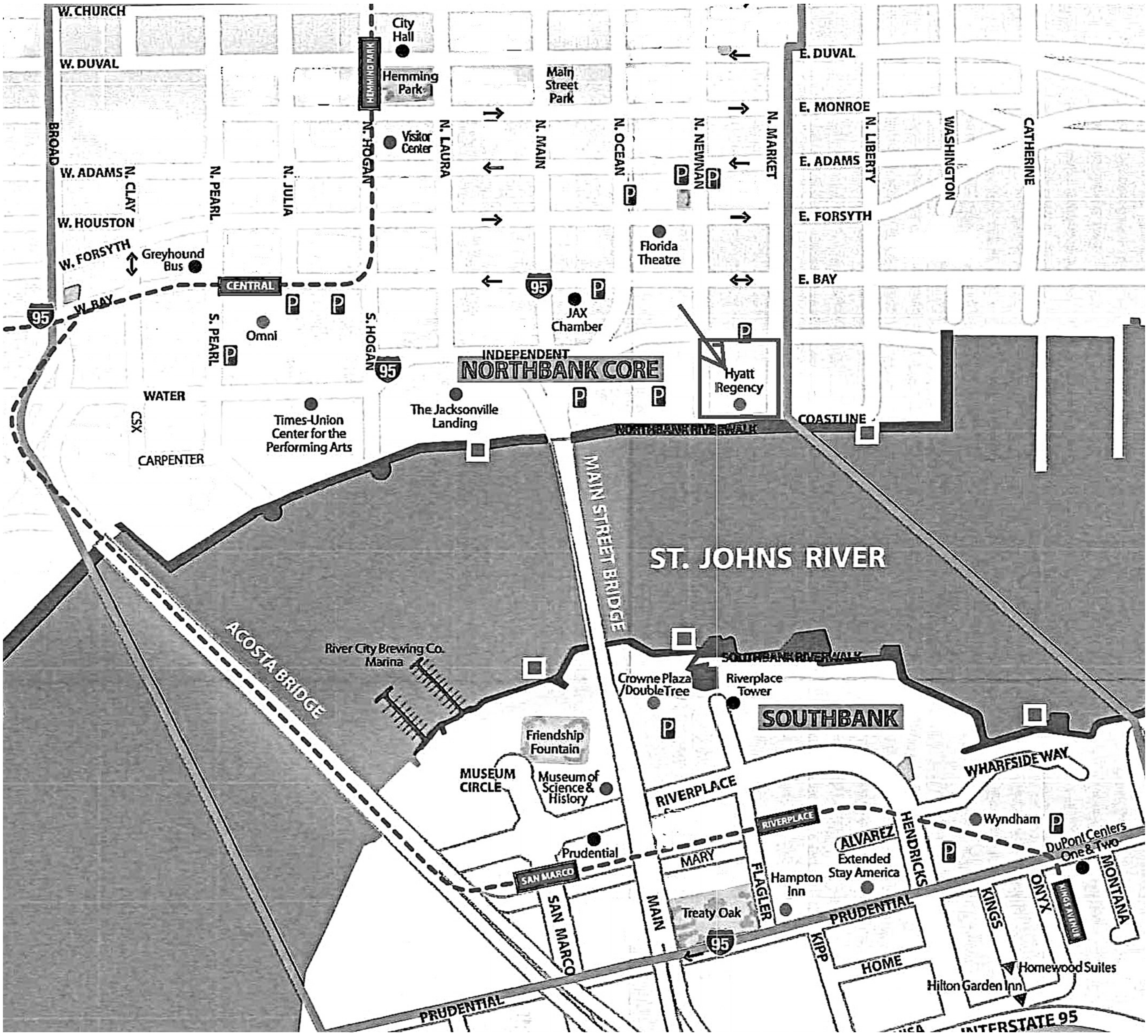


Third Floor



Fourth Floor




W. CHURCH
W. DUVAL
City Hall
HEMMING PARK
Hemming Park
Main Street Park
E. DUVAL
E. MONROE
E. ADAMS
E. FORSYTH
E. BAY
BROAD
W. ADAMS
N. CLAY
N. PEARL
N. JULIA
N. HOGAN
Visitor Center
N. LAURA
N. MAIN
N. OCEAN
N. NEWNAN
N. MARKET
N. LIBERTY
WASHINGTON
CATHERINE
W. HOUSTON
W. FORSYTH
Greyhound Bus
Florida Theatre
CENTRAL
W. BAY
95
JAX Chamber
S. PEARL
Omni
S. HOGAN
INDEPENDENT
NORTHBANK CORE
Hyatt Regency
WATER
CSX
CARPENTER
Times-Union Center for the Performing Arts
The Jacksonville Landing
NORTHBANK RIVERWALK
COASTLINE
ST. JOHNS RIVER
MAIN STREET BRIDGE
ACOSTA BRIDGE
River City Brewing Co. Marina
SOUTHBANK RIVERWALK
Crowne Plaza /DoubleTree
Riverplace Tower
SOUTHBANK
Friendship Fountain
MUSEUM CIRCLE
Museum of Science & History
RIVERPLACE
WHARFSIDE WAY
Wyndham
Prudential
SAN MARCO
MARY
ALVAREZ
Extended Stay America
HENDRICKS
DuPont Centers One & Two
MONTANA
ONYX
KINGS
KINGS AVENUE
SAN MARCO
MAIN
Treaty Oak
FLAGLER
Hampton Inn
PRUDENTIAL
KIPP
HOME
Homewood Suites
Hilton Garden Inn
PRUDENTIAL
INTERSTATE 95

TECHNICAL PROGRAM CALENDER
170th Meeting
Jacksonville, Florida
2–6 November 2015

MONDAY MORNING

9:00 1aAB **Animal Bioacoustics and Psychological and Physiological Acoustics:** Comparative Neurophysiology of the Auditory System I: Session in Honor of Albert Feng. City Terrace 9

8:30 1aAO **Acoustical Oceanography, Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics:** Acoustics of High Latitude Oceans I. River Terrace 2

8:30 1aNS **Noise:** Environmental and Community Noise. Grand Ballroom 1

8:35 1aPA **Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics:** Phononic Metamaterials I. St. Johns

8:30 1aSP **Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics:** Direction of Arrival (DOA) Estimation, Source Localization, Classification, and Tracking Using Small Aperture Arrays I. City Terrace 7

MONDAY AFTERNOON

2:00 1pAA **Architectural Acoustics:** Sustainability and Acoustics. Grand Ballroom 3

1:30 1pAB **Animal Bioacoustics and Psychological and Physiological Acoustics:** Comparative Neurophysiology of the Auditory System II: Session in Honor of Albert Feng. City Terrace 9

1:15 1pAO **Acoustical Oceanography, Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics:** Acoustics of High Latitude Oceans II. River Terrace 2

1:30 1pBA **Biomedical Acoustics:** Medical Ultrasound and Imaging. Clearwater

1:00 1pNS **Noise and Animal Bioacoustics:** Soundscape and Its Application. Grand Ballroom 1

1:00 1pPA **Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics:** Phononic Metamaterials II. St. Johns

1:30 1pSP **Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography:** Direction of Arrival (DOA) Estimation, Source Localization, Classification, and Tracking Using Small Aperture Arrays II. City Terrace 7

MONDAY EVENING

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Sonic Booms: A "Super" Sonic Saga. Grand Ballroom 6

TUESDAY MORNING

8:00 2aAA **Architectural Acoustics and Noise:** Acoustics of Multifamily Dwellings. Grand Ballroom 3

9:00 2aAB **Animal Bioacoustics:** Bioacoustics Across Disciplines: Detecting and Analyzing Sounds. City Terrace 9

8:15 2aAO **Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics:** Passive-Acoustic Inversion Using Sources of Opportunity I. River Terrace 2

8:00 2aBA **Biomedical Acoustics and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications I. Clearwater

8:00 2aEAa **Engineering Acoustics:** Vector Sensors: Theory and Applications. Orlando

10:15 2aEAb **Engineering Acoustics:** Analysis of Sound Sources. Orlando

8:00 2aED **Education in Acoustics and Musical Acoustics:** Effective and Engaging Teaching Methods in Acoustics. Grand Ballroom 6

8:30 2aNS **Noise:** Damage Risk Criteria for Noise Exposure I. Grand Ballroom 1

8:35 2aSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Flow-Induced Vibration. Daytona

8:30 2aSCa **Speech Communication:** Speech Production Potpourri (Poster Session). Grand Ballroom 8

10:30 2aSCb **Speech Communication:** Analysis and Processing of Speech Signals (Poster Session). Grand Ballroom 8

9:00 2aSP **Signal Processing in Acoustics:** Detection, Feature Recognition, and Communication. City Terrace 7

TUESDAY AFTERNOON

1:20 2pAAa **Architectural Acoustics and Musical Acoustics:** Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio I. Grand Ballroom 3

2:55 2pAAb **Architectural Acoustics and Noise:** Measuring Sound Fields in Healthcare Environments. Grand Ballroom 2

1:00 2pABa **Animal Bioacoustics:** Bioacoustics Across Disciplines: Emitting Sound. City Terrace 9

3:30 2pABb **Animal Bioacoustics:** Bioacoustics (Poster Session). Grand Ballroom 8

1:30 2pAO **Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics:** Passive-Acoustic Inversion Using Sources of Opportunity II. River Terrace 2

1:00 2pBA **Biomedical Acoustics and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications II. St. Johns

1:30 2pEA **Engineering Acoustics:** Analysis of Sound Sources, Receivers, and Attenuators. Orlando

1:30 2pED **Education in Acoustics:** Take 5's. Daytona

3:15 2pID **Interdisciplinary:** Guidance from the Experts: Applying for Grants and Fellowships. Daytona

1:00 2pNS **Noise:** Damage Risk Criteria for Noise Exposure II. Grand Ballroom 1

1:10 2pPA **Physical Acoustics:** Acoustic Characterization of Critical Phenomena. Clearwater

1:00 2pSA **Structural Acoustics and Vibration, Noise, and Architectural Acoustics:** Structural Acoustics and Vibration in Buildings. Grand Ballroom 7

1:30 2pSCa **Speech Communication:** Forensic Acoustics, Speaker Identification, and Voice. Grand Ballroom 6

3:30 2pSCb **Speech Communication:** Speech Perception Potpourri (Poster Session). Grand Ballroom 8

WEDNESDAY MORNING

8:00 3aAAa **Architectural Acoustics and Musical Acoustics:** Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio II. Grand Ballroom 3

9:35 3aAAb **Architectural Acoustics:** Worship Space Acoustics: Three Decades of Design. Grand Ballroom 2

10:55 3aAO **Acoustical Oceanography:** Munk Award Lecture. River Terrace 2

8:00 3aBAa **Biomedical Acoustics:** Sonothrombolysis. Clearwater

11:15 3aBAb **Biomedical Acoustics:** Therapeutic Ultrasound, Microbubbles, and Bioeffects I. Clearwater

8:00 3aEA **Engineering Acoustics:** Test Facilities and Acoustic Calibration. Orlando

10:00 3aED **Education in Acoustics:** Undergraduate Research Exposition (Poster Session). Grand Ballroom Foyer

8:30 3aNS **Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics:** Role of Fit-Testing Systems for Hearing Protection Devices. Grand Ballroom 1

8:00 3aPA **Physical Acoustics:** General Topics In Physical Acoustics I. St. Johns

8:00 3aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics Potpourri (Poster Session). Grand Ballroom 8

8:00 3aSA **Structural Acoustics and Vibration, Signal Processing in Acoustics, and Physical Acoustics:** Nonlinear Techniques for Nondestructive Evaluation. Daytona

9:00 3aSC **Speech Communication:** Various Topics in Speech Communication. Grand Ballroom 6

9:00 3aSP **Signal Processing in Acoustics:** Random Matrix Theory in Acoustics and Signal Processing. City Terrace 7

7:45 3aUW **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Animal Bioacoustics:** 50 Years of Underwater Acoustics under ASA. River Terrace 2

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics:** AIA CEU Course Presenters Training Session. Grand Ballroom 3

1:00 3pBA **Biomedical Acoustics:** Therapeutic Ultrasound, Microbubbles, and Bioeffects II. Clearwater

2:10 3pED **Education in Acoustics:** Acoustics Education Prize Lecture. Grand Ballroom 6

1:00 3pID **Interdisciplinary:** Hot Topics in Acoustics. Grand Ballroom 2

1:45 3pSC **Speech Communication:** Speech Style and Sociophonetics (Poster Session). Grand Ballroom 8

THURSDAY MORNING

8:55 4aAA **Architectural Acoustics and Noise:** Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance 1. Grand Ballroom 3

8:30	4aAB	**Animal Bioacoustics:** Avian Bioacoustics. City Terrace 9
8:30	4aBA	**Biomedical Acoustics and Physical Acoustics:** Numerical and Analytical Modeling of Medical Ultrasound I. Clearwater
8:30	4aEA	**Engineering Acoustics:** Acoustic Material Characterization Methods. Orlando
9:00	4aMU	**Musical Acoustics:** Stick-Slip Processes in Musical Instruments. Grand Ballroom 2
8:30	4aNS	**Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics:** Thoughts on the Next Generation of ANSI Loudness Standards. Grand Ballroom 1
8:30	4aPA	**Physical Acoustics and Noise:** Launch Vehicle Acoustics I: Acoustics of Launch Vehicles and Supersonic Jets. St. Johns
8:50	4aSC	**Speech Communication:** Development of Speech Production and Perception Across the Lifespan. Grand Ballroom 6
8:00	4aUW	**Underwater Acoustics and Signal Processing in Acoustics:** Environmental Variability Impact on Shallow Water Acoustics I. River Terrace 2

THURSDAY AFTERNOON

1:00	4pAAa	**Architectural Acoustics and Noise:** Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance II. Grand Ballroom 3
3:20	4pAAb	**Architectural Acoustics:** Architectural Acoustics Potpourri. Grand Ballroom 3
1:00	4pAB	**Animal Bioacoustics and Acoustical Oceanography:** Bioacoustics Research In Latin America. City Terrace 9
1:30	4pBA	**Biomedical Acoustics and Physical Acoustics:** Numerical and Analytical Modeling of Medical Ultrasound II. Clearwater
1:00	4pEA	**Engineering Acoustics and Structural Acoustics and Vibration:** Layered Media. Orlando
1:00	4pMU	**Musical Acoustics:** Acoustical Evolution of Musical Instruments. Grand Ballroom 2
1:15	4pNS	**Noise, Physical Acoustics, and Animal Bioacoustics:** Wind Turbine Noise. Grand Ballroom 1
1:35	4pPA	**Physical Acoustics and Noise:** Launch Vehicle Acoustics II: Analysis and Modeling of Noise from Supersonic Jets. St. Johns
1:30	4pPP	**Psychological and Physiological Acoustics:** Pitch, Loudness, and Other Perceptual Phenomena. Grand Ballroom 7
2:45	4pSA	**Structural Acoustics and Vibration:** Novel Treatments in Vibration Damping. Daytona
2:00	4pSCa	**Speech Communication and Signal Processing in Acoustics:** Advancing Methods for Analyzing Dialect Variation. Grand Ballroom 6
4:05	4pSCb	**Speech Communication:** Contributions to the Special Sessions (Poster Session). Grand Ballroom 8
1:30	4pSP	**Signal Processing in Acoustics:** Algorithm, Analysis, and Beamforming. City Terrace 7
1:00	4pUW	**Underwater Acoustics and Signal Processing in Acoustics:** Environmental Variability Impact on Shallow Water Acoustics II. River Terrace 2

FRIDAY MORNING

8:20	5aAB	**Animal Bioacoustics:** New Discoveries in Bat Vocal Communication. City Terrace 9
9:00	5aAO	**Acoustical Oceanography and Underwater Acoustics:** Topics in Acoustical Oceanography. City Terrace 7
8:30	5aMU	**Musical Acoustics:** General Topics in Musical Acoustics. Grand Ballroom 2
9:00	5aNS	**Noise:** Noise Potpourri. Grand Ballroom 1
8:00	5aPA	**Physical Acoustics:** General Topics in Physical Acoustics II. Clearwater
8:30	5aSA	**Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration. Orlando
8:30	5aSCa	**Speech Communication:** Intonation, Tone, and Prosody (Poster Session). Grand Ballroom 8
10:30	5aSCb	**Speech Communication:** Foreign Accent and Multilingual Speech Production and Perception (Poster Session). Grand Ballroom 8
8:00	5aUW	**Underwater Acoustics:** Communications, Transducers, Target Response, and Nonlinear Acoustics. River Terrace 2

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 2 Nov, 7:30 a.m.	Executive Council	Room 4108
Mon, 2 Nov, 3:30 p.m.	Technical Council	Room 4108
Tue, 3 Nov, 7:00 a.m.	Archives & History	Boardroom 2
Tue, 3 Nov, 7:00 a.m.	ASA Press Editorial Board	Boardroom 1
Tue, 3 Nov, 7:00 a.m.	POMA Editorial Board	Boardroom 3
Tue, 3 Nov, 7:30 a.m.	Panel on Public Policy	City Terrace 12
Tue, 3 Nov, 11:45 a.m.	Editorial Board	River Terrace 1
Tue, 3 Nov, 12:00 noon	Activity Kit	Boardroom 2
Tue, 3 Nov, 12:00 noon	Student Council	Room 4108
Tue, 3 Nov, 12:30 p.m.	Prizes & Special Fellowships	Boardroom 3
Tue, 3 Nov, 1:30 p.m.	Meetings	River Terrace 3
Tue, 3 Nov, 4:00 p.m.	Books+	Boardroom 1
Tue, 3 Nov, 4:30 p.m.	Newman Fund Advisory	Boardroom 3
Tue, 3 Nov, 5:00 p.m.	Women in Acoustics	River Terrace 3
Wed, 4 Nov, 6:45 a.m.	International Research & Education	Boardroom 2
Wed, 4 Nov, 7:00 a.m.	College of Fellows	Boardroom 3
Wed, 4 Nov, 7:00 a.m.	Publication Policy	City Terrace 12
Wed, 4 Nov, 7:00 a.m.	Regional Chapters	River Terrace 3
Wed, 4 Nov, 11:00 a.m.	Medals and Awards	Room 4108
Wed, 4 Nov, 11:30 a.m.	Public Relations	Boardroom 1
Wed, 4 Nov, 12:00 noon	Membership	Boardroom 3
Wed, 4 Nov, 1:30 p.m.	AS Foundation Board	Boardroom 2
Wed, 4 Nov, 5:00 p.m.	Education in Acoustics	Clearwater
Wed, 4 Nov, 5:00 p.m.	Acoustics Today Advisory	Room 4108
Thu, 5 Nov, 7:30 a.m.	Tutorials	Boardroom 1
Thu, 5 Nov, 7:30 a.m.	Investment	Boardroom 2
Thu, 5 Nov, 2:00 p.m.	Publishing Services	Boardroom 2
Thu, 5 Nov, 4:30 p.m.	External Affairs	Boardroom 1
Thu, 5 Nov, 4:30 p.m.	Internal Affairs	Boardroom 3
Fri, 6 Nov, 7:00 a.m.	Technical Council	Room 4108
Fri, 6 Nov, 11:00 a.m.	Executive Council	Room 4108

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 3 Nov, 4:30 p.m.	Engineering Acoustics	Orlando
Tue, 3 Nov, 7:30 p.m.	Acoustical Oceanography	River Terrace 2
Tue, 3 Nov, 7:30 p.m.	Animal Bioacoustics	City Terrace 9
Tue, 3 Nov, 7:30 p.m.	Architectural Acoustics	Grand Ballroom 3
Tue, 3 Nov, 7:30 p.m.	Physical Acoustics	St. Johns
Tue, 3 Nov, 7:30 p.m.	Psychological and Physiological Acoustics	Grand Ballroom 7
Tue, 3 Nov, 7:30 p.m.	Structural Acoustics and Vibration	Daytona
Wed, 4 Nov, 7:30 p.m.	Biomedical Acoustics	Clearwater
Wed, 4 Nov, 8:00 p.m.	Signal Processing in Acoustics	City Terrace 7
Thu, 5 Nov, 7:30 p.m.	Musical Acoustics	Grand Ballroom 1
Thu, 5 Nov, 7:30 p.m.	Noise	Grand Ballroom 2
Thu, 5 Nov, 7:30 p.m.	Speech Communication	Grand Ballroom 6
Thu, 5 Nov, 7:30 p.m.	Underwater Acoustics	River Terrace 2

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 2 Nov, 5:00 p.m.	S12/WG27-Outdoor Measurements	Boardroom 3
Mon, 2 Nov, 7:00 p.m.	ASACOS Steering	Boardroom 2
Tue, 3 Nov, 7:30 a.m.	ASACOS	Room 4108
Tue, 3 Nov, 3:00 p.m.	S3/SC1/WG6-Testing Toothed Whale Hearing	Boardroom 2

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 2-5 Nov 7:30 a.m. - 5:00 p.m. Fri, 6 Nov 7:30 a.m. - 12:00 noon	Registration	Grand Ballroom Foyer
Mon-Thu, 2-5 Nov, 7:00 a.m. - 5:00 p.m. Fri, 6 Nov, 7:00 a.m. - 12:00 noon	E-mail	Boardroom 4
Mon-Thu, 2-5 Nov, 7:00 a.m. - 5:00 p.m. Fri, 6 Nov 8:00 a.m. - 12:00 noon	Internet Zone	City Terrace 11
Mon-Thu, 2-5 Nov 7:00 a.m. - 5:00 p.m. Fri, 6 Nov 7:00 a.m. - 12:00 noon	A/V Preview	Boardroom 4
Mon-Thu, 2-5 Nov 8:00 a.m. - 10:00 a.m.	Accompanying Persons	Room 4104
Mon-Thu, 2-5 Nov, 8:00 a.m. - 5:00 p.m.	Gallery of Acoustics	Grand Ballroom Foyer
Sun, 1 Nov, 1:00 p.m. - 5:00 p.m. Mon, 2 Nov 8:30 a.m. -12:30 p.m.	Short Course	Daytona
Mon-Fri, 2-6 Nov 9:45 a.m. - 10:30 a.m.	A.M. Coffee Break	Grand Ballroom Foyer
Tue-Thu, 3-5 Nov 12:00 p.m. - 1:00 p.m.	Resume Help Desk	Grand Ballroom Foyer
Mon, 2 Nov 5:00 p.m. - 5:30 p.m.	New Student Orientation	City Terrace 9
Mon, 2 Nov 5:30 p.m. - 6:45 p.m.	Student Meet and Greet	River Terrace 3
Mon, 2 Nov, 5:30 p.m. - 7:00 p.m.	Exhibit Opening Reception	Grand Ballroom Foyer
Tue, 3 Nov, 9:00 a.m. - 5:00 p.m.	Exhibit	Grand Ballroom Foyer
Tue, 3 Nov, 2:45 p.m. - 3:30 p.m.	Afternoon Break	Grand Ballroom Foyer-Exhibit
Tue, 3 Nov, 6:00 p.m. - 7:30 p.m.	Social Hour	Grand Ballroom 4/5
Wed, 4 Nov, 8:45 a.m. - 11:00 a.m.	Technical Tour - Florida Theatre	Meet in hotel lobby at 8:45 a.m.
Wed, 4 Nov, 9:00 a.m. - 12:00 noon	Exhibit	Grand Ballroom Foyer
Wed, 4 Nov, 11:30 a.m. - 1:30 p.m.	Women in Acoustics Luncheon	River Terrace 1
Wed, 4 Nov, 3:30 p.m.	Annual Membership Meeting	Grand Ballroom 4
Wed, 4 Nov, 3:30 p.m. - 5:00 p.m.	Plenary Session/Awards Ceremony	Grand Ballroom 4
Wed, 4 Nov, 6:00 p.m. - 8:00 p.m.	Student Reception	River Terrace 1
Wed, 4 Nov, 8:00 p.m. - 12:00 midnight	ASA Jam	Grand Ballroom 4
Thu, 5 Nov, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Grand Ballroom 4
Thu, 5 Nov, 6:00 p.m. - 7:30 p.m.	Social Hour	River Terrace 1 River Deck 1

170th Meeting of the Acoustical Society of America

The 170th meeting of the Acoustical Society of America will be held Monday through Friday, 2–6 November 2015 at the Hyatt Regency Jacksonville Riverfront Hotel, Jacksonville, Florida.

SECTION HEADINGS

1. HOTEL INFORMATION

The Hyatt Regency Jacksonville Riverfront Hotel is the headquarters hotel where all meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact the Hyatt Regency Jacksonville Riverfront Hotel, 225 East Coastline Drive, Jacksonville, FL 32202 for information about room availability.

2. TRANSPORTATION AND TRAVEL

Jacksonville International Airport (JAX) provides non-stop flights to more than 30 major U.S. cities and is only 15 minutes from downtown Jacksonville. The airport offers 100 daily arrivals and departures on all major domestic air carriers and a network of regional carriers. More information about Jacksonville International Airport is available at FlyJax.

Airport transportation alternatives include: Taxis (15-minute ride, $30 to $40 USD); Shared-ride service: (SuperShuttle, $24 per person one way, $48 round-trip, 800-258-3826, SuperShuttle.com); Rental Car: All major companies are represented in the airport; Bus service: AirJTA (CT3) operates seven days a week and it is the most economical trip from downtown to the airport. For more information, visit the airport courtesy desk or visit JTA's website to plan your trip. All ground transportation services, including rental car agencies, taxi and limousine services, are located on the lower level of the terminal building adjacent to baggage claim. **Safety Advisory:** Ignore offers of ground transportation from solicitors inside the terminal building. While these solicitors may look and sound official, they are not authorized to solicit passengers.

Bus and Train Transportation: Amtrak (Amtrak.com) and Greyhound (greyhound.com) both serve Jacksonville. The Amtrak station is located at 3570 Clifford Lane, Jacksonville, 6.5 miles from the Hyatt Regency Jacksonville Riverfront. The Greyhound Station is located at 10 N. Pearl Street, Jacksonville and is 0.64 miles from the hotel.

Getting Around Jacksonville: **The Water Taxi** for travel between the north and south bank of the St. Johns River. Board a taxi for scenic and serene trips without waiting longer than 20 minutes at the pickup points. **Trolley**: The Riverside Trolley, which links Riverside and downtown and the Beach Trolley, which provides access throughout the beach neighborhoods. Please visit, jtafla.com, for schedules as the Trolley only operates during specific times of the week and year. **Hyatt Shuttle Service:** The Hyatt's complimentary hotel shuttle provides dependable transit to locations within three miles of the Hyatt. Take advantage of this service Monday through Friday from 6:00 a.m. to 10:00 a.m. and again from 5:00 p.m. to 9:00 p.m.

Parking: Hyatt Regency Jacksonville Riverfront offers a parking deck managed and operated by ProPark that provides direct access into the hotel. Options include overnight, self, and valet parking. Guests may use their hotel key card to enter and exit the garage after validating with the Front Desk. The garage entrance is located off of Newnan Street. Rates are; USD $15 for 24 hour self-parking, USD $20 for 24 hour valet parking. Hourly self-parking is USD $6 per hour. Garage clearance: 6' 7". Additional public parking is available in the nearby area, rates vary.

3. STUDENT TRANSPORTATION SUBSIDIES

To encourage student participation, limited funds are available to defray partially the cost of travel expenses of students to attend Acoustical Society meetings. Instructions for applying for travel subsidies are given in the Call for Papers which can be found online at http://acousticalsociety.org. The deadline for the present meeting has passed but this information may be useful in the future.

4. MESSAGES FOR ATTENDEES

A message board will be located in the Grand Ballroom Foyer near the ASA registration desk. Check the board during the week as messages may be posted by attendees who do not have cell phone numbers of other attendees.

5. REGISTRATION

Registration is required for all attendees and accompanying persons. Registration badges must be worn in order to participate in technical sessions and other meeting activities.

Registration will open on Monday, 2 November at 7:30 a.m. in the Grand Ballroom Foyer on the second floor (see floor plan on page A10).

Checks or travelers checks in U.S. funds drawn on U.S. banks and Visa, MasterCard and American Express credit cards will be accepted for payment of registration. Meeting attendees who have pre-registered may pick up their badges and registration materials at the pre-registration desk.

The registration fees (in USD) are $570 for members of the Acoustical Society of America; $645 for non-members, $150 for Emeritus members (Emeritus status pre-approved by ASA), $295 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $100 for ASA Student members, $140 for students who are not members of ASA, $115 for Undergraduate Students, and $150 for accompanying persons.

One-day registration is available at $295 for members and $360 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions). A nonmember who pays the $645 nonmember registration fee and simultaneously applies for Associate Membership in the Acoustical Society of America will be given a $50 discount off their dues payment for 2016 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the registration fee, but nonmember invited speakers who participate in the meeting only on the day of their presentation may register without charge. The registration fee for nonmember invited speakers who wish to participate for more than one day is $115 and includes a one-year Associate Membership in the ASA upon completion of an application form.

Special note to students who pre-registered online: You will also be required to show your student id card when picking-up your registration materials at the meeting.

6. ASSISTIVE LISTENING DEVICES

The ASA has purchased assistive listening devices (ALDs) for the benefit of meeting attendees who need them at technical sessions. Any attendee who will require an assistive listening device should advise the Society in advance of the meeting by writing to: Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@acousticalsociety.org

7. TECHNICAL SESSIONS

The technical program includes 90 sessions with 742 abstracts scheduled for presentation during the meeting.

A floor plan of the Hyatt Hotel appears on page A10. Session Chairs have been instructed to adhere strictly to the printed time schedule, both to be fair to all speakers and to permit attendees to schedule moving from one session to another to hear specific papers. If an author is not present to deliver a lecture-style paper, the Session Chairs have been instructed either to call for additional discussion of papers already given or to declare a short recess so that subsequent papers are not given ahead of the designated times.

Several sessions are scheduled in poster format, with the display times indicated in the program schedule.

8. TECHNICAL SESSION DESIGNATIONS

The first character is a number indicating the day the session will be held, as follows:

1-Monday, 2 November
2-Tuesday, 3 November
3-Wednesday, 4 November
4-Thursday, 5 November
5-Friday, 6 November

The second character is a lower case "a" for a.m., "p" for p.m., or "e" for evening corresponding to the time of day the session will take place. The third and fourth characters are capital letters indicating the primary Technical Committee that organized the session using the following abbreviations or codes:

AA Architectural Acoustics
AB Animal Bioacoustics
AO Acoustical Oceanography
BA Biomedical Acoustics
EA Engineering Acoustics
ED Education in Acoustics
ID Interdisciplinary
MU Musical Acoustics
NS Noise
PA Physical Acoustics
PP Psychological and Physiological Acoustics
SA Structural Acoustics and Vibration
SC Speech Communication
SP Signal Processing in Acoustics
UW Underwater Acoustics

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case "a" or "b" is used to distinguish the sessions. Each paper within a session is identified by a paper number following the session-designating characters, in conventional manner. As hypothetical examples: paper 2pEA3 would be the third paper in a session on Tuesday afternoon organized by the Engineering Acoustics Technical Committee; 3pSAb5 would be the fifth paper in the second of two sessions on Wednesday afternoon sponsored by the Structural Acoustics and Vibration Technical Committee.

Note that technical sessions are listed both in the calendar and the body of the program in the numerical and alphabetical order of the session designations rather than the order of their starting times. For example, session 3aAA would be listed ahead of session 3aAO even if the latter session begins earlier in the same morning.

9. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 4 November, at 1:00 p.m. in Grand Ballroom 2. Papers will be presented on current topics in the fields of Acoustical Oceanography, Animal Bioacoustics and Underwater Acoustics.

10. ROSSING PRIZE IN ACOUSTICS EDUCATION AND ACOUSTICS EDUCATION PRIZE LECTURE

The 2015 Rossing Prize in Acoustics Education will be presented to Yang-Hann Kim, Korean Advanced Institute for Science and Technology (KAIST), at the Plenary Session on Wednesday, 4 November. Yang-Hann Kim will present the Acoustics Education Prize Lecture titled "Propagation of acoustic education in space and time" on Wednesday, 4 November, at 2:10 p.m. in Session 3pED in Grand Ballroom 6.

11. WALTER MUNK AWARD AND MUNK AWARD LECTURE

Carl Wunsch, Cecil and Ida Green Professor of Physical Oceanography, Emeritus, Massachusetts Institute of Technology and Visiting Professor of Physical Oceanography at Harvard University has been named recipient of the 2015 Walter Munk Award for Distinguished Research in Oceanography Related to Sound and the Sea. The Walter Munk Award is granted jointly by The Oceanography Society, the Office of Naval Research and the Office of the Oceanographer of the Navy.

Carl Wunsch will present the Munk Award Lecture on Wednesday, 4 November, at 10:55 a.m. in Session 3aAO in River Terrace 2. The award will be presented to Dr. Wunsch following the lecture and he will be introduced during the Plenary Session on Wednesday afternoon.

12. TUTORIAL LECTURE ON SONIC BOOMS

A tutorial presentation titled "Sonic Booms: A "Super" Sonic Saga" will be given by Victor W. Sparrow of Pennsylvania State University on Monday, 2 November, at 7:00 p.m. in Grand Ballroom 6.

This tutorial will provide an introduction to sonic booms, including background, current status of research, and future prospects. Aircraft manufacturers have plans to build new civilian supersonic passenger aircraft, and these new aircraft intend to have sonic booms that are much quieter than those produced by either military aircraft or the now-retired Concorde supersonic airliner. If the aircraft manufacturers meet their stated goals, millions of people around the world will potentially begin hearing quiet sonic booms in the future.

In conjunction with the Tutorial Lecture on sonic booms on Monday evening, Gulfstream Aerospace Corporation will be bringing their sonic boom simulator to the Jacksonville, FL meeting. The Supersonic Acoustic Signature Simulator II is a specially equipped mobile audio booth designed to accurately reproduce the noise an observer on the ground would hear if a supersonic aircraft flew by. More specifically, the visitor at the ASA meeting will experience a back-to-back comparison of two radically different synthesized sonic booms. The first sonic boom signature will represent the "traditional" N-wave signature produced by the Concorde. The second sonic boom will represent a shaped signature that is representative of a low-boom aircraft. The intent of this demonstration is to provide the visitor with an opportunity to experience a fully immersive simulation that contains a sophisticated auralization of stimuli that contribute to subjective response caused by a sonic boom.

The simulator only will be available on Monday, November 2 (12:00 noon to 7:00 p.m.), and Tuesday, November 3 (9:00 a.m. to 6:00 p.m.), of the Jacksonville meeting. The simulator will be located at Jacksonville Landing, just steps away from the Hyatt Regency. Walking directions will be provided on site via convenient signs. Please come and take advantage of this special opportunity to hear the Supersonic Acoustic Signature Simulator II.

Lecture notes will be available at the tutorial in limited supply; only preregistrants will be guaranteed receipt of a set of notes.

The registration fee is USD $25 (USD $12 for students with current student IDs).

13. SHORT COURSE ON FINITE ELEMENTS IN ACOUSTICS

A short course on Finite Elements in Acoustics will be given in two parts: Sunday, 1 November, from 1:00 p.m. to 5:00 p.m. and Monday, 2 November, from 8:30 a.m. to 12:30 p.m. in the Daytona Room.

This short course will provide an overview of the theory and application of finite element methods (FEM) in acoustics. The instructor is Jeffrey Cipolla, an Associate Principal at WAI, who has over 20 years of experience in engineering mechanics, numerical simulations, project planning and management.

Instructional materials and coffee breaks are included. Onsite registration at the meeting will be on a space-available basis.

14. UNDERGRADUATE RESEARCH POSTER EXPOSITION

The Undergraduate Research Exposition will be held Wednesday morning, 4 November, 10:00 a.m. to 12:00 noon in session 3aED in the Grand Ballroom Foyer. The 2015 Undergraduate Research Exposition is a forum for undergraduate students to present their research pertaining to any area of acoustics and can also include overview papers on undergraduate research programs, designed to inspire and foster growth of undergraduate research throughout the Society. It is intended to encourage undergraduates to express their knowledge and interest in acoustics and foster their participation in the Society. Four awards, up to $500 each, will be made to help undergraduates with travel costs associated with attending the meeting and presenting a poster.

Please visit this poster session to view the work of the undergraduates and to encourage them in their work in acoustics.

15. EXHIBIT AND EXHIBIT RECEPTION

An instrument and equipment exhibit conveniently located near the registration area and meeting rooms, will be located in the Grand Ballroom Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems and other exhibits on acoustics.

The Exhibit will open on Monday evening at 5:30 p.m. with an evening reception with lite snacks and a complimentary drink. Coffee breaks on Tuesday and Wednesday mornings (9:45 a.m. to 10:30 a.m.) will be held in the exhibit area as well as an afternoon break on Tuesday (2:45 p.m. to 3:30 p.m.).

Exhibit hours are Monday, 2 November, 5:30 p.m. to 7:00 p.m., Tuesday, 3 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 November, 9:00 a.m. to 12:00 noon.

16. RESUME HELP DESK

Are you interested in applying for graduate school, a postdoctoral opportunity, a research scientist position, a faculty opening, or other position involving acoustics? If you are, please stop by the ASA Resume Help Desk in the Grand Ballroom foyer near the ASA registration desk. Members of the ASA experienced in hiring will be available review at your CV, cover letter, and research and teaching statements to provide tips and suggestions to help you most effectively present yourself in today's competitive job market. The ASA Resume Help Desk will be staffed on Tuesday, Wednesday, and Thursday during the lunch hour (12:00 noon to 1:00 p.m.) for walk-up meetings. Appointments during these three lunch hours will also be available via a sign-up sheet.

17. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at the Hyatt. The meetings on Tuesday and Thursday will be held in the evenings after the socials, except Engineering Acoustics which will meet at 4:30 p.m. on Tuesday. On Wednesday, Biomedical Acoustics will meet at 7:30 p.m. and Signal Processing in Acoustics will meet at 8:00 p.m. The schedule and rooms for each Committee meeting are given on page A8.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussions.

18. TECHNICAL TOUR

Wednesday, 4 November, 8:45 a.m. to 11:00 a.m. Tour fee: USD $10 (the collected fees will be donated to the Florida Theatre).

A tour of the historic Florida Theatre (http://floridatheatre.com) in Jacksonville will be led by Kenny Stubblefield. The Florida Theatre, an architectural and cultural treasure, opened in 1927 and is listed on the National Register of Historic Places. It has hosted vaudeville, silent films, ballet, rock-n-roll, etc. The Theatre is located 0.28 miles from the Hyatt Regency Jacksonville Riverfront Hotel. Participants will meet in the registration area of the hotel lobby at 8:45 a.m. for the quick walk to the Theatre.

19. GALLERY OF ACOUSTICS

The Technical Committee on Signal Processing in Acoustics will sponsor the 16th Gallery of Acoustics at the meeting. Its purpose is to enhance ASA meetings by providing a setting for researchers to display their work to all meeting attendees in a forum emphasizing the diversity, interdisciplinary, and artistic nature of acoustics. The Gallery of Acoustics provides a means by which we can all share and appreciate the natural beauty, aesthetic, and artistic appeal of acoustic phenomena: This is a forum where science meets art. A cash prize of USD $500 will be awarded to the winning entry.

The Gallery will be held in the Grand Ballroom Foyer, Monday through Thursday, 2-5 November, from 8:00 a.m. to 5:00 p.m.

Meeting attendees are asked to vote on the entries on the basis of aesthetic/artistic appeal, ability to convey and exchange information, and originality. Ballots will be included in registration envelopes or distributed to onsite registrants at the meeting.

A ballot box will be available at the Gallery where ballots should be deposited by Thursday, 5 November, at 5:00 p.m.

20. ANNUAL MEMBERSHIP MEETING

The Annual Membership Meeting of the Acoustical Society of America will be held at 3:30 p.m. on Wednesday, 4 November 2015, in Grand Ballroom 4 at the Hyatt Regency Jacksonville Riverfront Hotel, 225 East Coastline Drive, Jacksonville, FL 32202.

21. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 4 November, at 3:30 p.m. in Grand Ballroom 4.

The Rossing Prize in Acoustics Education will be presented to Yang-Hann Kim. The Silver Medal in Engineering Acoustics will be presented to John L. Butler, the Silver Medal in Psychological and Physiological Acoustics will be presented to Roy D. Patterson, the Silver Medal in Signal Processing in Acoustics will be presented to Brian G. Ferguson, and the Silver Medal in Speech Communication will be presented to John J. Ohala.

Certificates will be presented to Fellows elected at the Pittsburgh meeting of the Society. See page 1851 for a list of fellows.

Carl Wunsch, recipient of the Munk Award will be introduced and science writing awards will be presented to Trevor Cox and Kelly Servick.

All attendees are welcome and encouraged to attend. Please join us to honor and congratulate these medalists and other award recipients.

22. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will not be held at the Jacksonville meeting.

Meetings of selected advisory working groups are often held in conjunction with Society meetings and are listed in the Schedule of Committee Meetings and Other Events on page A16 or on the standards bulletin board in the registration area, e.g., S12/WGI8-Room Criteria.

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: Susan Blaeser, ASA Standards Manager, Standards Secretariat, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; T.: 631-390-0215; F: 631-923-2875; E: asastds@acousticalsociety.org

23. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:45 a.m. to 10:30 a.m. Breaks on Monday, Thursday and Friday will be held in Grand Ballroom Foyer.

Morning breaks on Tuesday and Wednesday will be held in the Grand Ballroom Foyer in the Exhibit area. There will also be an afternoon break on Tuesday from 2:45 p.m. to 3:30 p.m. in the Exhibit area.

24. A/V PREVIEW ROOM

Boardroom 4 on the third floor will be set up as an A/V preview room for authors' convenience, and will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 12:00 noon.

25. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Jacksonville meeting will have a published proceedings, and submission is optional. The proceedings will be a separate volume of the online journal, "Proceedings of Meetings on Acoustics" (POMA). This is an open access journal, so that its articles are available in pdf format for downloading without charge to anyone in the world. Authors who are scheduled to present papers at the meeting are encouraged to prepare a suitable version in pdf format that will appear in POMA. It is not necessary to wait until after the meeting to submit one's paper to POMA. Further information regarding POMA can be found at the site http://acousticsauthors.org. Published papers from previous meeting can be seen at the site http://asadl/poma.

26. E-MAIL AND INTERNET ZONE

Wi-Fi will be available in all ASA meeting rooms and spaces.

Computers providing e-mail access will be available 7:00 a.m. to 5:00 p.m., Monday to Thursday and 7:00 a.m. to 12:00 noon on Friday in Boardroom 4.

Tables with power cords will be set up in City Terrace 11 for attendees to gather and to power-up their electronic devices.

27. SOCIALS

Socials will be held on Tuesday evening, 6:00 p.m. to 7:30 p.m. in Grand Ballroom 4 and 5 and Thursday evening, 6:00 p.m. to 7:30 p.m. in River Terrace1/River Deck1. The ASA hosts these social hours to provide a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field. A second goal of the socials is to provide a sufficient meal so that meeting attendees can attend the open meetings of Technical Committees that begin immediately after the socials

28. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture will be held on Thursday, 5 November, at 12:00 noon in Grand Ballroom 4. The luncheon is open to all attendees and their guests. The speaker is Dr. Joseph Travis, Robert O. Lawton Distinguished Professor at Florida State University. Dr. Travis is a renowned field biologist. His research focuses on how ecological processes drive evolutionary ones. Dr. Travis's lecture is entitled, "Approaches to science, falsification vs. hypothetico-deductive, and some of the consequences of failing to recognize the limitations of each." Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 4 November. The cost is $30.00 per ticket.

29. STUDENTS MEET MEMBERS FOR LUNCH

The ASA Education Committee arranges for a student to meet one-on-one with a member of the Acoustical Society over lunch. The purpose is to make it easier for students to meet and interact with members at ASA Meetings. Each lunch pairing is arranged separately. Students who are interested should contact Dr. David Blackstock, University of Texas at Austin, by email dtb@mail.utexas.edu. Please provide your name, university, department, degree you are seeking (BS, MS, or PhD), research field, acoustical interests, your supervisor's name, days you are free for lunch, and abstract number (or title) of any paper(s) you are presenting. The sign-up deadline is 12 days before the start of the Meeting, but an earlier sign-up is strongly encouraged. Each participant pays for his/her own meal.

30. STUDENT EVENTS: NEW STUDENTS ORIENTATION, MEET AND GREET, STUDENT RECEPTION, GRANT AND FELLOWSHIP PANEL

Follow the student twitter throughout the meeting @ASAStudents.

A New Students Orientation will be held from 5:00 p.m. to 5:30 p.m. on Monday, 2 November, in City Terrace 9 for all students to learn about the activities and opportunities available for students at the Jacksonville meeting. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in River Terrace 3. Refreshments and a cash bar will be available. Students are encouraged to attend the tutorial lecture on Monday which begins at 7:00 p.m. in Grand Ballroom 6.

A Fellowship and Grant Panel titled Guidance from the Experts: Applying for Grants and Fellowships, will be held in session 2pID on Tuesday, 3 November, from 3:15 p.m. to 4:45 p.m. in the Daytona Room. The Panel will include successful fellowship winners, selection committee members, and fellowship agency members organized by the Student Council. The panelists will briefly introduce themselves and answer questions regarding grant and fellowship opportunities and application advice.

The Students' Reception will be held on Wednesday, 4 November, from 6:00 p.m. to 8:00 p.m. in River Terrace 1. This reception, sponsored by the Acoustical Society of America and supported by the National Council of Acoustical Consultants, will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

Students will find a sticker in their registration envelopes to place on their name tags identifying them as students. Although wearing the sticker is not mandatory, it will allow for easier networking between students and other meeting attendees.

Students are encouraged to refer to the student guide, also found in their envelopes, for important program and meeting information pertaining only to students attending the ASA meeting.

They are also encouraged to visit the official ASA Student Home Page at http://asastudentcouncil.org// to learn more about student involvement in ASA.

31. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 4 November, in River Terrace 1 on the third floor of the Hyatt. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 3 November. The fee is USD$30 for non-students and USD$15 for students.

32. JAM SESSION

You are invited to Ballroom 4 on Wednesday night, 4 November, from 8:00 p.m. to midnight for the ASA Jam. Bring your axe, horn, sticks, voice, or anything else that makes music. Musicians and non-musicians are all welcome to attend. A full PA system, backline equipment, guitars, bass, keyboard, and drum set will be provided. All attendees will enjoy live music, a cash bar with snacks, and all-around good times. Don't miss out.

Earplugs donated by Etymotic Research Inc. will be available at the Jam.

33. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Jacksonville meeting. The on-site registration fee for accompanying persons is USD$150. A hospitality room for accompanying persons will be open in Room 4104 at the Hyatt from 8:00 a.m. to 10:00 a.m. Monday through Thursday. This entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

Downtown Jacksonville is the business, cultural and entertainment center of Northeast Florida. The St. Johns River, one of a few rivers in the United States that flows north, is the crowning jewel offering visitors and residents some of the most picturesque views. Downtown is also home to a large variety of theatres and performing arts venues. There are also a myriad of museums, tours, and river cruises for you to enjoy! Visit www.visitjacksonville.com for additional information.

Just steps from the Hyatt Regency Jacksonville Riverfront is Jacksonville Landing offering, shops, restaurants and entertainment. For more information visit Jacksonville Landing.

34. WEATHER

Jacksonville's climate averages of 221 sunny days per year. November highs average at 74 during the day with lows in the 50s at night.

35. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Judy R. Dubno and Catherine L. Rogers, Cochairs; David P. Knobles, Acoustical Oceanography; Benjamin Taft, Animal Bioacoustics; David T. Bradley and Damian J. Doria, Architectural Acoustics; Siddhartha Sikdar, Biomedical Acoustics; Andrew Piacsek and Michelle Vigeant, Education in Acoustics; Roger T. Richards and Kenneth M. Walsh, Engineering Acoustics; Andrew C.H. Morrison, Musical Acoustics; Eric L. Reuter and Victor W. Sparrow, Noise; Michael R. Haberman, Physical Acoustics; Christopher A. Brown, Psychological and Physiological Acoustics; Said Assous and Ning Xiang, Signal Processing in Acoustics; Alexander L. Francis, Speech Communication; Robert M. Koch, Structural Acoustics and Vibration; Nicholas Chotiros, Underwater Acoustics; Matthew Kamrath, Student Council.

36. MEETING ORGANIZING COMMITTEE

Richard H. Morris, Chair; Judy R. Dubno and Catherine L. Rogers, Technical Program Cochairs; Anand Swaminathan and Derek Jones, Audio-Visual; Kaitlin Lansford, Volunteer Coordination; Richard H. Morris, Technical Tour; Gail Paolino, Meeting Administrator.

37. PHOTOGRAPHING AND RECORDING

Photographing and recording during regular sessions are not permitted without prior permission from the Acoustical Society.

38. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2015 issue of *The Journal of the Acoustical Society of America*. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the *Journal*.

39. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

See the guidelines for computer projection in section 41 below.

- Allow at least one minute of your talk for each slide (e.g., PowerPoint). No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Generally, too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.

- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institutions logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.

Presentation

- Organize your talk with introduction, body, and summary or conclusion. Include only ideas, results, and concepts that can be explained adequately in the allotted time. Four elements to include are:
- Statement of research problem
- Research methodology
- Review of results
- Conclusions
- Generally, no more than 3–5 key points can be covered adequately in a 15-minute talk so keep it concise.
- Rehearse your talk so you can confidently deliver it in the allotted time. Session Chairs have been instructed to adhere to the time schedule and to stop your presentation if you run over.
- An A/V preview room will be available for viewing computer presentations before your session starts. It is advisable to preview your presentation because in most cases you will be asked to load your presentation onto a computer which may have different software or a different configuration from your own computer.
- Arrive early enough so that you can meet the session chair, load your presentation on the computer provided, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation. There will be many presenters loading their materials just prior to the session so it is very important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They won't have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

40. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

The poster should be centered around two or three key points supported by the title, figures, and text. The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.

- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
- Background
- Objective, purpose, or goal
- Hypotheses
- Methodology
- Results (including data, figures, or tables)
- Discussion
- Implications and future research
- References and Acknowledgment

Design and layout

- A board approximately 8 ft. wide × 4 ft. high will be provided for the display of each poster. Supplies will be available for attaching the poster to the display board. Each board will be marked with an abstract number.
- Typically posters are arranged from left to right and top to bottom. Numbering sections or placing arrows between sections can help guide the viewer through the poster.
- Centered at the top of the poster, include a section with the abstract number, paper title, and author names and affiliations. An institutional logo may be added. Keep the design relatively simple and uncluttered. Avoid glossy paper.

Lettering and text

- Font size for the title should be large (e.g., 70-point font)
- Font size for the main elements should be large enough to facilitate readability from 2 yards away (e.g., 32 point font). The font size for other elements, such as references, may be smaller (e.g., 20–24 point font).
- Sans serif fonts (e.g., Arial, Calibri, Helvetica) are much easier to read than serif fonts (e.g., Times New Roman).
- Text should be brief and presented in a bullet-point list as much as possible. Long paragraphs are difficult to read in a poster presentation setting.

Visuals

- Graphs, photographs, and schematics should be large enough to see from 2 yards (e.g., 8 × 10 inches).
- Figure captions or bulleted annotation of major findings next to figures are essential. To ensure that all visual elements are "stand alone," axes should be labeled and all symbols should be explained.
- Tables should be used sparingly and presented in a simplified format.

Presentation

- Prepare a brief oral summary of your poster and short answers to likely questions in advance.
- The presentation should cover the key points of the poster so that the audience can understand the main findings. Further details of the work should be left for discussion after the initial poster presentation.
- It is recommended that authors practice their poster presentation in front of colleagues before the meeting. Authors should request feedback about the oral presentation as well as poster content and layout.

Other suggestions

- You may wish to prepare reduced-size copies of the poster (e.g., 8 1/2 × 11 sheets) to distribute to interested audience members.

41. GUIDELINES FOR USE OF COMPUTER PROJECTION

A PC computer with monaural audio playback capability and projector will be provided in each meeting room on which all authors who plan to use computer projection should load their presentations. Authors should bring computer presentations on a CD or USB drive to load onto the provided computer and should arrive at the meeting rooms at least 30 minutes before the start of their sessions. Assistance in loading presentations onto the computers will be provided.

Note that only PC format will be supported so authors using Macs must save their presentations for projection in PC format. Also, authors who plan to play audio during their presentations should insure that their sound files are also saved on the CD or USB drive.

Introduction

It is essential that each speaker who plans to use his/her own laptop connect to the computer projection system in the A/V preview room prior to session start time to verify that the presentation will work properly. Technical assistance is available in the A/V preview room at the meeting, but not in session rooms. Presenters whose computers fail to project for any reason will not be granted extra time.

Guidelines

- Set your computer's screen resolution to 1024x768 pixels or to the resolution indicated by the AV technical support. If it looks OK, it will probably look OK to your audience during your presentation.
- Remember that graphics can be animated or quickly toggled among several options: Comparisons between figures may be made temporally rather than spatially.
- Animations often run more slowly on laptops connected to computer video projectors than when not so connected. Test the effectiveness of your animations before your assigned presentation time on a similar projection system (e.g., in the A/V preview room). Avoid real-time calculations in favor of pre-calculation and saving of images.
- If you will use your own laptop instead of the computer provided, connect your laptop to the projector during the question/answer period of the previous speaker. It is good protocol to initiate your slide show (e.g., run PowerPoint) immediately once connected, so the audience doesn't have to wait. If there are any problems, the session chair will endeavor to assist you, but it is your responsibility to ensure that the technical details have been worked out ahead of time.• During the presentation have your laptop running with main power instead of using battery power to insure that the laptop is running at full CPU speed. This will also guarantee that your laptop does not run out of power during your presentation.

SPECIFIC HARDWARE CONFIGURATIONS

Macintosh

Older Macs require a special adapter to connect the video output port to the standard 15-pin male DIN connector. Make sure you have one with you.

- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that your monitor controls are set to mirroring.

If it's an older PowerBook, it may not have video mirroring, but something called simulscan, which is essentially the same.

- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configuration, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

PC

- Make sure your computer has the standard female 15-pin DE-15 video output connector. Some computers require an adaptor.
- Once your computer is physically connected, you will need to toggle the video display on. Most PCS use either ALT-F5 or F6, as indicated by a little video monitor icon on the appropriate key. Some systems require more elaborate keystroke combinations to activate this feature. Verify your laptop's compatibility with the projector in the A/V preview room. Likewise, you may have to set your laptop's resolution and color depth via the monitor's Control Panel to match that of the projector, which settings you should verify prior to your session.

Linux

- Most Linux laptops have a function key marked CRT/LCD or two symbols representing computer versus projector. Often that key toggles on and off the VGA output of the computer, but in some cases, doing so will cause the computer to crash. One fix for this is to boot up the BIOS and look for a field marked CRT/LCD (or similar). This field can be set to Both, in which case the signal to the laptop is always presented to the VGA output jack on the back of the computer. Once connected to a computer projector, the signal will appear automatically, without toggling the function key. Once you get it working, don't touch it and it should continue to work, even after reboot.

42. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; Telephone: 516-576-2360; Fax: 631-923-2875; E-mail: asa@acousticalsociety.org

171st Meeting, Salt Lake City, Utah, 23–27 May 2016
172nd Meeting, Honolulu, Hawaii, 28 November–2 December 2016
(The 5th Joint meeting of the Acoustical Society of America and the Acoustical Society of Japan)
173rd Meeting, Boston, Massachusetts, 25–29 June 2017
(The 3rd joint meeting of the Acoustical Society of America and the European Acoustics Association
174th Meeting, New Orleans, Louisiana, 4–8 December 2017
175th Meeting, Minneapolis, Minnesota, 7–11 May 2018
176th Meeting, Victoria, Canada, 6–9 November 2018
177th Meeting, Louisville, Kentucky, 13–17 May 2019
178th Meeting, TBD
179th Meeting, Chicago, Illinois, spring 2020
180th Meeting, Cancun, Mexico, fall 2020

APPLICATIONS DUE 22 FEBRUARY 2016

PHYSICAL ACOUSTICS SUMMER SCHOOL

PASS 2016

The 2016 Physical Acoustics Summer School, PASS 2016, will be convened at Daniels Summit Lodge, Heber City Utah, south of Salt Lake City Utah and Park City Utah, with arrival on 15 June and departure on 21 June. This is the same venue as was for the 2014 Physical Acoustics Summer School. Participation, including Students, Lecturers, and Discussion Leaders, is limited to fifty. Full-time participation of all is required. Part-timers and visitors are not permitted.

PURPOSE: The purpose of the Summer School is to bring graduate Students, distinguished Lecturers, and Discussion Leaders together to discuss a wide variety of subjects in physical acoustics. PASS will give students opportunities to meet experts and discuss topics they would not ordinarily encounter in their own colleges and universities.

STUDENTS: The focus of PASS is on intermediate and advanced graduate students.

LECTURERS AND DISCUSSION LEADERS: The Organizing Committee will select approximately eight lecture topics and appropriate Lecturers and Discussion Leaders.

COSTS: Participants provide their own transportation to and from Salt Lake City Airport (SLC). Some van travel assistance from and to SLC will be available. There is a $200 Student Registration Fee. The other costs of participation, including room and board, based on multiple-occupancy for Students, will be paid by sponsors.

PROGRAM: Program information, included in a full Preliminary Schedule, will be available to all those who request the Application Forms from the webpage (below) at the National Center for Physical Acoustics (NCPA) at The University of Mississippi.

APPLICATION DEADLINE: Completed Application Forms for the 2016 Summer School must be received at NCPA no later than Monday 22 February 2016.

APPLICATIONS: All participants must have a completed Application Form on file. Copies of the Application Form and a Preliminary Schedule are available at http://ncpa.olemiss.edu/pass-2016/. PASS 2016 contacts are Debra A. Perrier at National Center for Physical Acoustics, The University of Mississippi, P.O. Box 1848, 145 Hill Drive, University MS 38677-1848 or at (662) 915-5840, Fax (662) 915-7494, dperrier@olemiss.edu and J. R. (Josh) Gladden, PASS 2016 Director and NCPA Director, at the same address or at (662) 915-7428, Fax (662) 915-7494, jgladden@olemiss.edu.

ASA School 2016

Living in the Acoustic Environment

21-22 May 2016
Snowbird, Utah

ASA School 2016 is an Acoustical Society of America event where graduate students and early career acousticians in all areas of acoustics can learn about and discuss a wide variety of topics related to the interdisciplinary acoustical theme *Living in the Acoustic Environment*. **ASA School 2016** follows on the success of ASA School 2012 and ASA School 2014 and will provide opportunities for meeting faculty and fellow students, discussing research topics, developing collaborations and professional relationships within acoustics, and mentoring.

Program and Costs

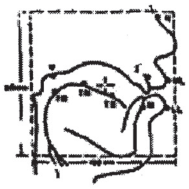
ASA School 2016 will take place at Snowbird, a mountain resort near Salt Lake City, Utah. Lectures, demonstrations, and discussions will be given by distinguished acousticians in a two-day program covering topics in Animal Bioacoustics, Architectural Acoustics, Engineering Acoustics, Musical Acoustics, Noise, Psychological and Physiological Acoustics, and Speech Communication. The registration fee is $50. Hotel rooms at Snowbird for two nights (double-occupancy), meals, course materials, and transportation from Snowbird to the ASA meeting location in Salt Lake City at the close of ASA School 2016 are provided by ASA. Participants are responsible for their own travel costs and arrangements including transportation to Snowbird.

Participants and Requirements

ASA School 2016 is targeted to graduate students in all areas of acoustics and early career acousticians (within 3 years of terminal degree). Attendance is limited to 60 participants who are expected to attend all School events and the ASA meeting immediately following on 23-27 May 2016. ASA School attendees are required to be an author on an abstract for presentation at the ASA Salt Lake City meeting.

Application and Deadlines

The application form and preliminary program will be available online in November, 2015 at www.AcousticalSociety.org.



NOW AVAILABLE ON DVD

MEASURING SPEECH PRODUCTION: VIDEO DEMONSTRATIONS OF SPEECH INSTRUMENTATION

This series of demonstrations, for use in teaching courses on speech acoustics, physiology and instrumentation are now available on DVD from the Acoustical Society of America. The DVD contains thirteen video demonstrations of equipment and techniques used in speech research. The demonstrations are categorized into three areas: (1) Respiration, phonation and aerodynamics; (2) Indirect articulatory measurements; (3) Direct articulatory measurements. A pdf file on the DVD describes the demonstrations and lists additional readings that are updated from the original videotape.

PART ONE - RESPIRATION, PHONATION AND AERODYNAMICS
1. The whole body plethysmograph in speech research. John J. Ohala
2. Aerodynamic end respiratory kinematic measures during speech. Elaine T. Stathopoulos
3. Physiologically based models of phonation. Ingo R. Titze
4. Use of the electroglottograph in the laboratory and clinic. James J. Mahshie
5. Endoscopy, stroboscopy, and transillumination in speech research. Anders Lofqvist, Kiyoshi Oshima

PART TWO - INDIRECT ARTICULATORY MEASUREMENTS
6. Magnetic resonance imaging (MRI) in speech research. Carol Gracco, Mark Tiede
7. Imaging the tongue with ultrasound. Maureen Stone
8. Estimating articulatory movement from acoustic data. Kenneth N. Stevens
9. Electromyography in speech research, Kiyoshi Oshima. Katherine S. Harris, Fredericka Bell-Berti

PART THREE - DIRECT ARTICULATORY MEASUREMENTS
10. The rise and fall of the soft palate: The Velotrace. Fredericka Bell-Berti, Rena A. Krakow, Dorothy Ross, Satoshi Horiguchi
11. Dynamic electropalatography. William J. Hardcastle, Fiona Gibbon
12. Measuring articulatory movements with an electromagnetic midsagittal articulometer (EMMA) system. Joseph S. Perkell, Mario A. Svirsky, Melanie L. Matthies, Joyce Manzella
13. Optoelectronic measurement of orofacial motions during speech production. Eric Vatikiotis-Bateton, Kevin Munhall, David Ostry

Each demonstration displays the instrument and how it is used; what the data look like; how data are analyzed and their applications for speech pathology, linguistics and speech processing. Anyone at any level interested in speech production and speech physiology will find these demonstrations useful. Price: $52.00

ORDER FORM

- Payment by Visa, MasterCard or American Express, or check or money order in US funds on US bank must accompany the order.
- Send orders to: Acoustical Society Publications, P.O. Box 1020, Sewickley PA 15143-9998; Tel: 412-741-1979; Fax: 412-741-0609
- Postage and handling: U.S. orders--$6 plus $2 for each additional DVD; Non-US orders: $10 plus $4 for each additional DVD.
- Returns are not accepted

Name____ [ ] ASA member [ ] Nonmember

Address____

City__________________ State/Prov__________ Zip______ Country____________________

Tel.:__________________ Fax:______________ Email:__________________________

Quantity		Unit Price	Total
_______	Measuring Speech Production DVD	$_______	$_________
	Shipping and Handling	$_______	$_________
		Total Remitted	$_________

[ ] Check or money order enclosed for $______________(U.S. funds/drawn on U.S. bank)

[ ] American Express [ ] VISA [ ] MasterCard

Account number:______________________________ Security Code:_________ Exp. Date:______________

Signature:____

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please use our secure web page to process your credit card payment (http://www.abdi-ecommerce10.com/asa) or securely fax this form to (516-576-2377).

Session 1aAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Comparative Neurophysiology of the Auditory System I: Session in Honor of Albert Feng

Andrea Simmons, Cochair
Brown University, Box 1821, Providence, RI 02912

Peter M. Narins, Cochair
Integrative Biology & Physiology, UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095

Chair's Introduction—9:00

Invited Papers

9:10

1aAB1. Sexual communication in ultrasonic frogs *Odorrana tormota* involves acoustic and various visual signals. Albert S. Feng (Dept. Molec. & Integrative Physiol., Univ. of Illinois, 524 Burrill Hall, Urbana, IL 61801, fengatcu@gmail.com), Fang Zhang, Pan Chen, Juan Zhao, and Zhuqing Chen (College of Life Sci., Anhui Normal Univ., Wuhu, China)

Female concave eared torrent frogs are known to produce courtship calls to attract males; yet, like other anurans, they also display positive phonotaxis toward male's calls. Thus, it is unclear how amplexus is formed and which sex makes the mating decision. We investigated how males and females interact and communicate during the reproductive season and found that females emitted vocalizations during the gravid state only. In playback experiments in the wild, female's courtship calls elicited numerous males to respond. Males' vocal responses were range dependent and accompanied by staccato calls and phonotaxis toward (but short of reaching) the loudspeaker. In short range encounters, gravid females are highly selective, and males must wait for female's invitational signal(s) for amplexus. Only a select male showing prominent dominant male's postures received invitational signal(s), comprising a variety of visual signals (head bob, belly inflation, toe tapping, and eye blink) and an "admission" acoustic signal. Upon receiving one or more of these signals, the male immediately hopped on the female's back to form amplexus. Our studies revealed that intersexual communication in concave eared torrent frogs is complex, involving both acoustic and visual signaling.

9:30

1aAB2. The spatio-temporal analysis of male-male vocal interactions in chorusing frogs. Douglas L. Jones, Russell L. Jones (Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Rama Ratnam (Coordinated Sci. Lab., Univ. of Illinois at Urbana-Champaign, 1308 W. Main St., Urbana, IL 61801, ratnam@illinois.edu)

In some chorusing frog species, such as the green treefrog (*Hyla cinerea*), males can rapidly adjust call-timing with reference to neighbors and avoid acoustic interference while maintaining call-rate. The rules underlying vocal interactions are largely unknown, presumably being species-specific and governed by acoustic and physiological factors. Here, we use a microphone-array technique that can simultaneously localize callers and selectively extract the calls of each frog. This allows us to analyze vocal interactions between individual callers at high spatial and temporal resolutions. We show that green treefrogs can synchronize with one another and prefer to time their calls antiphonally that are on average exactly at one-third and two-thirds of the inter-call intervals of a focal neighbor. When antiphonal calling is not possible, call collision is tolerated even though a continuum of phase positions are available. The communication system in green treefrogs is thus "discrete" with callers capable of rapidly switching between the discrete phase-slots (0, 1/3, and 2/3) in response to changes in a neighbor's phase. Further, call collision increases and phase-locking decreases, with increasing inter-caller spacing. We conclude that the vocal communication system in green treefrogs is capable of robust maintenance of inter-caller timing so as to maintain chorus synchrony.

9:50

1aAB3. Instantaneous reproductive isolation in polyploid treefrogs: Mechanisms and consequences. H. C. Gerhardt (Div. Biol. Sci., Univ. of Missouri, 215 Tucker Hall, Univ. Missouri, Columbia, MO 65211, gerhardth@missouri.edu)

Speciation by polyploidy—duplication of chromosome sets—commonly occurs in plants and happened twice in the early evolutionary history of vertebrates. The gray treefrog (*Hyla versicolor*) is a recently evolved tetraploid species found in the eastern United States, where it often occurs with a cryptic diploid species (*Hyla chrysoscelis*). Assortative mating by ploidy is favored by selection because of triploid hybrid sterility and is based solely on selective phonotaxis of females to the species (ploidy)-specific calls of males. The tetraploid species has arisen multiple times independently; the calls and preferences of frogs in these lineages are nearly the same. Autotriploids of the diploid species created in the laboratory show parallel shifts in both the key call-properties of males and female preferences

in the direction of the wild-type tetraploid species. These shifts are probably caused by changes in cell size (larger in tetraploids) and number (fewer in tetraploids) rather than the difference in gene dosage. The magnitude of changes expected in autotetraploids would be sufficient for instant pre-mating isolation of diploids and recently derived tetraploids, hence promoting instantaneous speciation, which would then be reinforced by selection against the production of infertile hybrids.

10:10–10:25 Break

10:25

1aAB4. Frogs, thalamotectal neurons, and other things I learned about from Al Feng. Daniel Llano (Univ. of Illinois at Urbana-Champaign, 2355 Beckman Inst., 405 N. Mathews, Urbana, IL 61801, d-llano@illinois.edu)

In this presentation, I will discuss our work on the thalamotectal system, work inspired Al Feng and others working in the anuran thalamus. Descending projections from the thalamus to inferior colliculus are evolutionarily highly conserved. Despite the long history of work on this pathway in anurans and other species, the basic organization of the mammalian auditory thalamotectal pathway has yet to be characterized. We therefore studied the basic organizational properties of this pathway in mouse. We observed that a large proportion of cells from the medial and paralaminar divisions of the auditory thalamus project to the lateral cortex of the inferior colliculus. In addition, the vast majority of thalamotectal cells are negative for calcium binding proteins typically found in thalamocortical neurons. Double tracer injections revealed that thalamotectal cells do not branch to the amygdala or cortex. Finally, unlike thalamocortical cells, thalamotectal cells do not show typical T-type calcium channel-dependent bursting. These data suggest that the thalamocollicular pathway is neurochemically, neurophysiologically and anatomically distinct from primary auditory thalamic pathways, and bears resemblances to the anuran thalamotectal system. These results will be discussed in the context of Al Feng's ongoing influence on our laboratory (including unusual items left behind in the lab…).

10:45

1aAB5. Explorations of the unique anuran ear: The contributions of Al Feng. Andrea M. Simmons (Cognit., Linguist & Psychol. Sci., Brown Univ., Providence, RI 02912, Andrea_Simmons@brown.edu)

The anuran inner ear is unique among vertebrate ears in containing three organs for sound detection—the amphibian papilla, the basilar papilla, and the saccule (a mixed auditory/vestibular organ). Al's anatomical, physiological, and behavioral work in several different anuran species, from the American bullfrog to the Chinese torrent frog, has significantly furthered our understanding of the evolution and operation of this unusual ear. In this talk, I will review this species diversity in inner ear systems and I will further discuss how this diversity emerges over ontogenetic (metamorphic) development. Following up on Al's research into the post-metamorphic development of the amphibian and basilar papillae in bullfrogs, I will present anatomical data on the larval development of the inner ear in two anuran species, the bullfrog and the African clawed frog, which are adapted to live in different biomes as adults. These results focus attention on how differences in post-metamorphic habitats are reflected in developmental timetables of the maturation of the ear during tadpole life.

11:05

1aAB6. Re-visiting Al Feng's contributions toward understanding the peripheral basis of acoustic communication in frogs. William Shofner (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, wshofner@indiana.edu)

Al has made substantial contributions toward understanding the peripheral auditory mechanisms underlying acoustic communication in frogs. Al was the first to demonstrate that the basilar papilla is innervated by high-frequency auditory nerve fibers, while the amphibian papilla gives rise to low- and mid-frequency fibers. As his graduate student, we subsequently showed that the tuning of these end organs changes during post-metamorphic development. Al developed a dorsal surgical approach to the auditory nerve, which allowed the coupling between the ears to be studied. He demonstrated that both the monaural directivity patterns and the coupling between ears are frequency dependent. In a follow-up study, we manipulated the coupling between ears and concluded that the frog's ear functions as a combination pressure/pressure gradient receiver. In this talk, I will re-visit the frequency-dependent crosstalk between the ears first described by Al in 1980. The crosstalk function is analyzed using an acoustic model of a tube with a side branch. This analysis shows the mouth cavity as the side branch is insufficient and yields a lowpass crosstalk function. Including the nares in parallel with the mouth cavity as a side branch yields a bandpass function for sound transmission that accounts for Al's empirical crosstalk data.

Session 1aAO

Acoustical Oceanography, Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics: Acoustics of High Latitude Oceans I

Aaron Thode, Cochair
SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Chair's Introduction—8:30

Invited Papers

8:35

1aAO1. The state of the Arctic Ocean, its variability and prediction—An overview. Wieslaw Maslowski (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, maslowsk@nps.edu)

The Arctic is a complex and integral part of the Earth system, influencing the global surface energy and moisture budgets, atmospheric and oceanic circulations, and geosphere-biosphere feedback. Some key influences are linked to the recent changes in the multi-year sea ice cover and the ocean. The ice cover is particularly important because it buffers air-sea heat fluxes and through ice-albedo feedback strongly influences Earth's absorption of solar radiation, especially by the ocean. Global warming has been most visibly manifested in the Arctic through a declining perennial sea ice cover, which has intensified during the late 1990s and the 2000s, resulting in record minima ice cover in 2007 and in 2012. This talk will provide an overview of the recent states and variability of the Arctic Ocean. We will focus on physical changes of potential relevance to the Arctic acoustics, including the past and present climatological changes, evolution of the upper ocean stratification and water masses, mesoscale processes, including eddies, mixing, coastal and boundary currents, and their linkages to the changing regime of the sea ice cover from multi-year to first-year sea ice. Finally, the latest advancements and outstanding challenges in modeling and prediction of arctic climate change will be discussed.

9:35

1aAO2. Airgun array sound measurements in the northeastern Chukchi Sea. David E. Hannay and Graham Warner (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, David.Hannay@jasco.com)

Underwater acoustic measurements were made during eight 2D/3D seismic surveys with large airgun arrays and nine shallow hazards surveys using small arrays (2–4 airguns) in the shallow (30–50 m) waters of the northeastern Chukchi Sea between 2006 and 2013. The acoustic measurements were made with calibrated seabed-deployed recorders to assess potential noise exposures to marine fauna. Data were acquired from directly under the airgun arrays to over 150 km distance. The seismic pulses contain signals above ambient noise between approximately 5 Hz and 15 kHz. Modal structure develops by about 1 km distance from the sources at frequencies between approximately 30 Hz and 600 Hz, but it is strongest from 100 Hz to 300 Hz. This ocean environment therefore acts as a frequency band-pass filter, selectively passing the intermediate sound frequencies to long distances. The modes are strongly dispersed, with low frequency components arriving up to 2 s later than the higher frequency sounds of the same mode. This causes received signals to have a down-swept spectral shape. We present sound level measurement results from all of the surveys and we discuss the frequency-dependent acoustic transmission loss in context with the modal propagation characteristics of this ocean environment.

10:00–10:15 Break

10:15

1aAO3. Unraveling the mystery of cold regime ecosystem variability in the Bering Sea through acoustics. Jennifer L. Miksis-Olds (Appl. Res. Lab., Penn State, PO Box 30, Mailstop 3510D, State College, PA 16804, jlm91@psu.edu) and Beth A. Stauffer (Biology, Univ. of Louisiana at Lafayette, Lafayette, LA)

The Bering Sea oscillates between warm and cold climatic regimes on 3–7 year cycles. During cold regimes, the bottom water layer, or cold pool, can remain below 2 °C the entire summer. When the cold pool forms, it acts as a cross-shelf barrier separating species of the outer shelf from others of the middle shelf and coastal areas. Backscatter recorded throughout the winter from echosounders on subsurface moorings in the southeastern Bering Sea provide a first glimpse on how dynamic the cold pool influence can be within cold regime years. The fall/winter scattering community composition was predictive of the structure during the first blooms the following spring. Seasonal environmental conditions were also observed to play a dominant role in summer lower trophic level dynamics. Delayed ice retreat in the summer of the coldest years was associated with increased abundance of large zooplankton; yet relatively warmer years during the same cold climatic regime yielded a shift to smaller zooplankton scatterers during summer. Chlorophyll concentrations showed varying levels of correlation to zooplankton patterns, and sparse cruise-based data suggested differences in phytoplankton community composition likely influenced these relationships. Data on phytoplankton community structure remains a desperately needed dataset to fully understand the ecosystem dynamics.

10:40

1aAO4. Summer and winter whales in the Pacific Arctic. Kathleen Stafford (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, stafford@apl.washington.edu)

Changes in sea ice phenology have been profound in the Pacific Arctic, where the seasonal open-water period has increased by ~1.5 months over the past 30 years. The greatest changes in the open water season have occurred in fall resulting in changes to the Arctic ecosystem, including increased primary productivity, changing food web structure, and opening of new habitat. In the "new normal" Arctic, sub-arctic "summer" whales (fin, humpback, and killer) are poised to inhabit new seasonally ice-free habitats in the Arctic. The spatial and seasonal occurrence of summer and "winter" (bowhead) whales over 5 years from September through December was examined by deploying hydrophones in Bering Strait and comparing acoustic occurrence of the species concomitant with decadal-scale changes in seasonal sea ice. Fin and humpback whale acoustic detections extended from summer to late autumn while killer whale detections were more sporadic. Inter-annual differences in acoustic detections appear to be driven by interannual differences in the environment. Bowhead whale detections generally began after the departure of the summer whales and continued through the winter. In a future with further seasonal sea ice reductions, however, increased competition for resources between sub-Arctic and Arctic species may arise to the detriment of winter whales.

11:05

1aAO5. Eight years of passive acoustic monitoring in the Alaskan Beaufort Sea: Lessons learned. Susanna B. Blackwell (Greeneridge Sci., Inc., 90 Arnold Pl., Ste. D, Santa Barbara, CA 93117, susanna@greeneridge.com), Christopher S. Nations (Western EcoSystems Technol., Cheyenne, WY), Aaron M. Thode (Scripps Inst. of Oceanogr., La Jolla, CA), and Katherine H. Kim (Greeneridge Sci., Inc., San Diego, CA)

Every year during open-water season in 2007–2014, up to 40 passive acoustic recorders with directional capability (DASARs) were deployed in the Beaufort Sea, as part of Shell's Marine Mammal Monitoring and Mitigation Program. The overarching goal of these deployments was to collect information on the effects of industrial operations, such as seismic exploration and drilling activities, on the behavior of bowhead whales during their fall migration. Recorders were placed on the continental shelf at depths of $<$55 m, offshore of the North Slope of Alaska, between Barter Island and Harrison Bay. Over the eight-year period, more than 3.1 million bowhead calls were localized. Concurrently, various types of industrial sounds were detected and quantified, such as airgun pulses or the tones produced by machinery. Analyses were then performed by matching—for each recorder—the number of calls localized with the amount of industrial sound detected. These analyses have shown that changes in calling behavior happen at low received levels of anthropogenic sound and lead to complex changes in calling behavior, that are governed by received sound levels, but also other factors, such as distance to the industrial activity or type of industrial sound. [Work sponsored by Shell Exploration and Production Company.]

Session 1aNS

Noise: Environmental and Community Noise

Eric L. Reuter, Chair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Contributed Papers

8:30

1aNS1. The initial development of a hybrid method for modeling outdoor sound propagation in urban areas. Matthew Kamrath, Julien Maillard, Philippe Jean, Dirk Van Maercke (Ctr. Scientifique et Technique du Bâtiment (CSTB), CSTB, 24 Rue Joseph Fourier, Saint-Martin-d'Hères 38400, France, matthew.kamrath@cstb.fr), and Judicaël Picaut (LUNAM Université, Ifsttar, AME, LAE, Bouguenais Cedex, France)

Accurately modeling urban outdoor sound propagation is a difficult problem. Using a frequency-domain or time-domain method is too expensive, and using an engineering, geometrical, or statistical method is too restrictive. For example, the ISO 9613-2, NMPB-Roads, and CNOSSOS-EU engineering methods only model diffraction for straight barriers. The Nord2000 and Harmonoise engineering methods can also model diffraction from wedge-shaped screens, but these modeling capabilities are still insufficient to model many potential noise mitigation solutions such as a gamma or T-shaped barrier. To extend the applicability of engineering methods, a detailed model like the boundary element method could characterize a complicated noise mitigation device versus a simple reference device. This presentation discusses the initial development of this hybrid method.

8:45

1aNS2. Predicted effects of meteorology on ground treatments. Shahram Taherzadeh (Eng. & Innovation, The Open Univ., Walton Hall, Milton Keynes MK7 6AA, United Kingdom, shahram.taherzadeh@open.ac.uk)

This presentation reports on calculations carried out to investigate the effect of meteorological conditions on the performance of ground treatments for traffic noise reduction. These treatments include placing strips of acoustically soft or rough surfaces near the source. The soft or rough strip can be raised up to 30 cm. In other words, the height of the rough elements can be as much as 30 cm. Their upper surface height can also be the same as the ground surface or, in other words, the treatments can be recessed. We present and discuss numerical predictions of the influence of downwind and upwind conditions on the insertion loss of raised and recessed, acoustically soft strips or strips of roughness elements in 2D using a Parabolic Equation code.

9:00

1aNS3. Hierarchical modeling approach to community noise annoyance. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army ERDC, 72 Lyme Rd, Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Nicole M. Wayant (Geospatial Res. Lab., U.S. Army ERDC, Alexandria, VA), Edward T. Nykaza (Construction Eng. Res. Lab., U.S. Army ERDC, Champaign, IL), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Chandler M. Armstrong (Construction Eng. Res. Lab., U.S. Army ERDC, Champaign, IL)

Previous efforts to predict community noise annoyance from objective measurements of sound exposure have been confounded by the large and unexplained scatter in survey data. We hypothesize that community noise annoyance can be rationally modeled and predicted through a hierarchical (multilevel) statistical paradigm, which incorporates variations in tolerance among individuals and communities. Simulations based on this conceptual model indicate that the individual- and community-level variations have distinct statistical signatures, both of which are clearly evident in existing meta-analyses of annoyance to transportation noise. We thus perform regression analyses of transportation noise annoyance using a multilevel, generalized linear model (GLM), which is statistically consistent with the conceptual model. The multilevel model strikes a compromise between no pooling and complete pooling of survey data, and enables the noise tolerances and their variations at the multiple model levels to be distinguished and quantified.

9:15

1aNS4. Classification of environmental noise sources using machine-learning methods. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil), Arnold P. Boedihardjo (ERDC-GRL, Alexandria, VA), Matthew G. Blevins, Andrew M. Hulva (ERDC-CERL, Champaign, IL), and Dan Valente (Chartbear, New York, NY)

Unattended and continuously running environmental noise monitoring systems can capture an intractable amount of data. The signals captured can include a multitude of sources (e.g., wind noise and anthropogenic noise sources) in addition to the environmental noise sources of interest (e.g., aircraft, vehicles, trains, and military weapons). In this presentation, we explore the use of machine-learning methods to effectively isolate and identify environmental noise sources captured on such a noise monitoring system. Specifically, we consider the use of both unsupervised (e.g., principle components analysis, clustering methods, and deep belief networks) and supervised (e.g., logistic regression, support vector machines, and neural networks) pattern-learning methods to derive the features of interest and classify the signals based on the obtained features. The generalization performance of each method is assessed using a dataset of over 120,000 human classified signals, and the strengths and weaknesses of each approach are discussed.

9:30

1aNS5. Using blind signal processing algorithms to remove wind noise from environmental noise assessments: A wind turbine amplitude modulation case study. Paul Kendrick, Sabine von Hünerbein, and Trevor J. Cox (Acoust. Res. Ctr., The Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, t.j.cox@salford.ac.uk)

Microphone wind noise can corrupt outdoor measurements and recordings. It is a particular problem for wind turbine measurements because these cannot be carried out when the wind speed is low. Wind shields can be used, but often the sound level from the turbine is low and even the most efficient shields may not provide sufficient attenuation of the microphone wind noise. This study starts by quantifying the effect that microphone wind noise has on the accuracy of two commonly used Amplitude Modulation (AM) metrics. A wind noise simulator and synthesized wind turbine sounds based on real measurements are used. The simulations show that even relatively low wind speeds of 3 m/s can cause large errors in the AM metrics. Microphone wind noise is intermittent, and consequently, one solution

is to analyze only uncorrupted parts of the recordings. This paper tests whether a single-ended wind noise detection algorithm can automatically find uncorrupted sections of the recording, and so recover the true AM metrics. Tests showed that doing this can reduce the error to ±2 dBA and ±0.5 dBA for the time and modulation-frequency domain AM metrics, respectively.

9:45–10:00 Break

10:00

1aNS6. Active noise control of rocket engine noise in an engine test facility for quiet nearby communities. Kwanhyung Lee and Yong Joe Kim (Texas A&M Univ., 1537 Pine Ridge Apt C, College Station, TX 77840, rhksgud92@tamu.edu)

The objective of this research project is to develop an Active Noise Control (ANC) system based on the Filtered-X Least Mean Square algorithm. The ANC system developed here will be installed in a rocket engine testing facility of SpaceX Company near McGregor, Texas. As the SpaceX Company's rocket engines have been increased in their powers, the engine noise from the testing facility is increasing rapidly. Although there is no noise complain currently from the nearby communities in McGregor, concerns are expected in the near future that the noise could possibly disturb people in McGregor. The ANC system can reduce the noise level using a few microphones and speakers connected to a DSP frontend, while a Passive Noise Control system would be much larger and require a higher cost to be implemented. This ANC system is currently implemented as a Single Input and Single Output system in a laboratory, effectively reducing the noise generated from a loudspeaker driven by the recorded rocket engine noise signal. Then, the system will be improved and operated as a Multiple Inputs and Multiple Outputs system to successfully adapt various wind, temperature, and humidity changes, leading to the maximum noise reduction toward McGregor.

10:15

1aNS7. Measurements and player surveys of crowd noise levels during college hockey ames. Brenna N. Boyd and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, bnboyd@unomaha.edu)

Measuring crowd noise in football stadiums has become popularized by the media in recent years, as teams have vied for the right to claim to have the "loudest stadium." Many fans believe that the louder they cheer, the more likely their team will win. While cheering gives players mental support, the extraneous noise could also be detrimental to effective communications on the field. Based on Barnard's previous study of crowd noise at outdoor football games (2011), the present study focuses on hockey arenas which, unlike football stadiums, typically have roofs that can add more surface area for sound to amplify. Measurements of the crowd noise levels and surveys of hockey players have been taken during four college hockey games within a closed-roof stadium. Results will be presented and discussed, with focus on determining the loudest section, if there is noise interference on the ice, and if there is correlation between noise level and goals acquired by the home team.

10:30

1aNS8. Road traffic noise mapping in the Federal University of Santa Maria, Brazil. Olmiro C. de Souza Neto (UFSM, Acampamento, 569, Santa Maria, Santa Maria 97050003, Brazil, olmirocz.eac@gmail.com) and Stephan Paul (Ctr. of Mobility Eng. and Lab. of Vib. and Acoust., UFSC, Joinville, Brazil)

The Federal University of Santa Maria is an academic community with teaching units, leisure and resting places, and a hospital. This different uses require that environmental noise limits, as given by international and national recommendations, are respected. Noise maps are one of the tools that can be used for soundscape evaluation and to check conformity with requirements. In this project, the data necessary for the development of the noise map of UFSM university campus have been collected, taking the traffic of motor vehicles on the campus as mayor noise source. A methodology for data collection was created based on the bibliography and input data required by the software. Noise maps, in terms of day sound pressure level isolines, were then generated by the software SoundPLAN 7.1. The noise map created shows that 35% of the teaching units are overexposed to road traffic noise and 45% of the students may be under negative effects of the road noise. It was also discovered that the most noise sensitive units, that are the hospital, a high school, and a day care center, have their facades exposed to day sound pressure levels higher than the recommendations.

10:45

1aNS9. Ordinal response model of community response. Chandler M. Armstrong and Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil)

Those analyzing community response to noise data often collapse ordinal scale data into a binomial variable to predict the percentage of the community that will be highly annoyed at given a noise dose. Collapsing the data simplifies the model, but also discards potentially useful information. An ordinal response model is one way to use all the information available in ordinal scale data. This paper compares the conventional binomial model to an ordinal response model to demonstrate the benefits and drawbacks of each. We have found that the ordinal response model may be a better option because some independent variables only influence response to noise at the lower end of the ordinal scale. The ordinal response model can detect effects across the entire ordinal range whereas the binomial model examines only the highly annoyed end of the scale.

11:00

1aNS10. Choosing noise dose bin widths when modeling community response to noise. Nicole Wayant (ERDC-GRL, Alexandria, VA), Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil), D. Keith Wilson (ERDC-CRREL, Hanover, NH), and Chandler M. Armstrong (ERDC-CERL, Champaign, IL)

While there are many studies on how communities respond to noise, there are few that explicitly define how the noise dose, used to fit community response models, are binned. When modeling the dose response relationship between noise and community annoyance, the choice of the noise bin width can drastically affect the community response or tolerance models fit to the data. To explore the optimal noise bin width, we fit various dose response models to bin widths between 1 and 10 dB and data that had been divided into bins of varying widths but an equal number of points in each bin. Additionally, we fit the models using a weighted regression procedure that weights each bin by its number of points, and find that this approach mitigates the modeling errors implicit to the (often arbitrary) noise bin width decision.

Session 1aPA

Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Phononic Metamaterials I

Joel Mobley, Chair
Physics and Astronomy, University of Mississippi, PO Box 1848, 1034 NCPA, University, MS 38677

Chair's Introduction—8:35

Invited Papers

8:40

1aPA1. Wave control with acoustic metasurfaces. Steven Cummer, Yangbo Xie, Wenqi Wang, and Bogdan-Ioan Popa (Duke Univ., PO Box 90291, Durham, NC 27708, cummer@duke.edu)

Acoustic metasurfaces are thin, engineered structures that can control the local reflection and transmission phase of acoustic waves, and thus enable a high degree of flexibility of wave manipulation. They are currently under active investigation by a multiple research groups for diffractive acoustic elements, surface mode excitation, extraordinary wave transmission, and a number of other possible applications. We will describe our recent work in the area of acoustic metasurfaces based on both rigid labyrinthine passive structures and piezoelectric-based active structures. Specific topics will include optimization of passive metasurface structure, nonlinear and reconfigurable active metasurfaces, and control of acoustic diffraction and surface waves with periodic metasurfaces.

9:00

1aPA2. Soft 3D acoustic metamaterials. Thomas Brunet, Olivier Poncelet, Christophe Aristégui (Université de Bordeaux, I2M, 351, cours de la libération, Bâtiment A4 - I2M/APY, Talence 33405, France, thomas.brunet@u-bordeaux.fr), Jacques Leng (Université de Bordeaux, LOF, Pessac, France), and Olivier Mondain-Monval (Université de Bordeaux, CRPP, Pessac, France)

One of the current challenges in the field of metamaterials is to extend beyond electromagnetism in other areas such as acoustics [1]. Soft matter techniques coupled with microfluidics provide a unique tool to take up this challenge because they allow for the production of 3D locally resonant (random) materials composed of various soft resonators [2]. The Mie (or cavity) resonators are interesting key elements for acoustic metamaterials since they may induce strong acoustic resonances provided large sound-speed contrasts between the inclusions and the host matrix [3]. For example, the use of "slow" (deformable) ferrofluid-droplets allows for the tuning of sharp Mie resonances, thanks to external magnetic fields [4]. For much higher sound-speed contrasts, we have recently shown that strong Mie resonances of "ultra-slow" beads made of soft porous silicone rubbers [5] could induce strong dispersion effects leading to materials with exotic (zero, negative) values of the acoustic index [6]. [1] Wegener, *Science* **342**, 939–940 (2013). [2] Brunet *et al.*, *Science* **342**, 323–324 (2013). [3] Li and Chan, *Phys. Rev. E* **70**, 055602 (2004). [4] Brunet *et al.*, *Phys. Rev. Letters* **111**, 264301 (2013). [5] Zimny *et al.*, *Langmuir* **31**, 3215–3221 (2015). [6] Brunet *et al.*, *Nature Mater.* **14**, 384–388 (2015).

9:20

1aPA3. Acoustic wave control with cylindrical metamaterial elements in water. Andrew Norris, Alexey S. Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu), and Michael Haberman (The Univ. of Texas at Austin, Austin, TX)

Devices based on transformation acoustics (TA) designed to control radiation of acoustic waves in water will be described. By restricting the TA mapping to a two-dimensional conformal transformation, it is shown that the properties in the physical device have constant density equal to the background density, while the effective acoustic medium must have spatially varying compressibility with values proportional to the area mapping. The talk will concentrate on the application of these ideas to a conformal TA device, the circle-to-square lens. A practical design procedure based on unit solid cylinder metamaterial elements will be described. The selection of elements is simplified using an Ashby-type chart of properties in terms of effective cylinder density vs. compressibility. Unlike previous gradient index lens devices of this type, the present design is almost optimally impedance and index matched. Analytical, numerical, and experimental results will be presented for a device with only 49 elements showing broadband focusing with positive gain in the preferred directions. [Work supported by ONR.]

9:40

1aPA4. Source driven homogenization and spatial dispersion effects in acoustic metamaterials. Caleb F. Sieck (Appl. Res. Labs. and Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, csieck@arlut.utexas.edu), Michael B. Muhlestein (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Andrea Alù (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Acoustic metamaterials (AMM) composed of dynamic subwavelength heterogeneities in a host fluid may generate an overall response that can be represented with dynamic effective parameters such as negative dynamic density or compressibility. Dynamic parameters imply that highly variable effective wavelengths exist even in the long wavelength limit where $k_0a << 1$, with k_0 representing the wavenumber in the host and a the descriptive size of the heterogeneity. The variability in effective wavelength is the result of strong frequency dispersion, often accompanied by nonlocal and spatial dispersion effects that complicate efforts to correctly homogenize the medium. This work presents a three-dimensional, source-driven, non-local homogenization scheme for a periodic AMM composed of a host fluid containing dynamic heterogeneities. The resulting constitutive relations couple macroscopic volume-strain and momentum fields and are analogous to the Willis relations of elastodynamics and bianisotropy in electromagnetism. The model accounts for first-order spatial dispersion effects in the long wavelength limit and reveals the origins of coupled field response. One dimensional examples of AMM will be used to demonstrate the homogenization procedure and the effects of spatial dispersion. [Work supported by ONR.]

10:00–10:30 Break

Contributed Papers

10:30

1aPA5. The acoustic magician hat: Broadband acoustic cloaking within a cavity with hard boundaries. Weiwei Kan (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Nanjing, Nanjing, China), Garcia-Chocano M. Victor, Francisco Cervera (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Valencia, Valencia, Spain), Bin Liang, Xin-ye Zhou, Lei-lei Yin (Dept of Phys., Univ Nanjing, Nanjing, China), Jianchun Cheng (State Key Lab. of Acoust., Chinese Acad. of Sci., Beijing, Beijing, China), and Jose Sanchez-Dehesa (Dept. of Electron. Eng., Universitat Politècnica de València, Wave Phenomena Group, Valencia, Valencia ES-46022, Spain, jsdehesa@upv.es)

This work reports the design, fabrication, and experimental validation of a broadband acoustic cloak for the concealing of three-dimensional (3D) objects placed inside an open cavity with arbitrary surfaces. This 3D cavity cloak represents the acoustic analogue of a magician hat, giving the illusion that a cavity with an object is empty. Transformation acoustics is employed to design this cavity cloak, whose parameters represent an anisotropic acoustic metamaterial. A practical realization is made of perforated layers fabricated by drilling subwavelength holes on 1-mm-thick Plexiglas plates. In both simulation and experimental results, concealing of the reference object by the device is shown for airborne sound with wavelengths between 10 cm and 17 cm.

10:45

1aPA6. Analysis of cloaks for flexural waves. Alfonso Climente (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Valencia, Valencia, Spain), Daniel Torrent (Universite de Bordeaux, Bordeaux, France), and Jose Sanchez-Dehesa (Dept. of Electron. Eng., Universidad Politecnica de Valencia, Wave Phenomena Group, Valencia, Valencia ES-46022, Spain, jsdehesa@upv.es)

We report a comprehensive analysis of the cloaks proposed for bending waves propagating in a thin metallic plate. The analysis uses a semi-analytical model which is able to reproduce the experimental data reported by Stenger and coworkers [see Phys. Rev. Lett., vol. 108, 014301 (2012)]. The model is based on the Kirkoff-Love equation of motion and employs the multilayer scattering algorithm to calculate the interaction of the propagating waves with the cloak. The boundary equations apply to the designed device give a complete description of the data without using simplified algorithms solved in a finite element framework. The performance of the cloak is characterized with the visibility factor, a parameter already employed in several acoustic cloaks. A discussion will be given of the performance of the cloak for flexural waves in comparison with acoustic cloaks. [Work supported by ONR.]

11:00

1aPA7. Reciprocity, passivity, and causality in media with coupled strain-momentum constitutive relations. Michael B. Muhlestein (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, mimuhle@gmail.com), Caleb F. Sieck, Andrea Alù (Appl. Res. Labs. and Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Metamaterials are heterogeneous materials and structures which, under certain circumstances, may be interpreted as a homogeneous material displaying extreme physical properties such as negative effective density or modulus. Acoustic metamaterials (AMM) therefore expand the parameter space for acoustic device design and are thus of interest for a wide range of applications. Most AMM are composed of linear and passive constituent materials. Any physically meaningful approximation of their overall response must therefore obey reciprocity, passivity, and causality. This requirement constrains the effective parameters and delineates the range of possible material properties for an AMM. While restrictions for standard constitutive equations in acoustics and elastodynamics are well known, they have not been explored in detail for media with coupled strain-momentum behavior which is required to fully describe AMM [Norris *et al.*, Proc. R. Soc. A, **467**, 1749–1769 (2011)]. This work derives the restrictions on coupled acoustic constitutive equations by requiring the overall response to be reciprocal, passive, and causal. The special cases of lossless and very low-loss materials are then considered and some useful approximate restrictions are also presented. Restrictions for the analogous case of generally anisotropic elastic strain-momentum coupled media will also be reported. [Work supported by ONR.]

11:15

1aPA8. Non-reciprocity and refraction of elastic waves in a solid with aligned parallel gaps. Xiaoshi Su and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Non-reciprocal and refractive devices for elastic waves are designed using a solid with aligned parallel gaps. The gaps are assumed to be thin so that they can be considered as parallel rectangular cracks separating effective thin plates. To formulate the transmission and reflection coefficients for SV- and P-wave, an analytical model is established using thin plate theory with displacement and force continuous at the junction between plate and exterior body. The analytical model compares well with full FEM simulations. The non-reciprocity effect for P-waves is achieved by sending an incident P-wave at a critical angle, at which total conversion to SV-wave

happens, with respect to the normal of a free boundary. An array of parallel gaps perpendicular to the propagation direction of the reflected waves stop the SV-wave but let P-wave travel through. Thus, the energy transmission is high from one side to another and is low from the opposite direction. The refractive effect for SV-waves is achieved by choosing the slope of the edge of plate array, and plate members for $|T| = 1$ at the same frequency with the ratio between plate length and flexural wavelength fixed. Examples of elastic non-reciprocal and wave steering devices will be discussed.

11:30

1aPA9. Acoustic metasurfaces for asymmetric transmission. Chen Shen and Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., 3131-I Walnut Creek Pkwy, Raleigh, NC 27606, cshen5@ncsu.edu)

We present a design of acoustic metasurfaces yielding highly asymmetric transmission within a certain frequency band. The design consists of a bottom layer of gradient-index metasurface and a top layer of low refractive index metasurface. When the incident waves transmit from the bottom side, the wave-front will be steered, and the total reflection occurs on the boundary of the top layer, leading to extremely low transmission. When the incident waves transmit from the top side, the transmission is high as the impedance of the low refractive index metasurface is matched with the background medium. Numerical simulations show that the transmission contrast between the two incident directions is high within a certain frequency band. Compared to previous designs, the proposed one can be more compact. This design may have potential applications in ultrasound imaging and noise reduction.

11:45

1aPA10. Design of one-way acoustic energy flow devices using Bayesian network classifiers. Benjamin M. Goldsberry, Stephanie G. Konarski, Timothy Klatt, and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmgoldsberry@gmail.com)

Linear acoustic metamaterials (AMM) displaying dynamic negative stiffness or density have been the subject of current research to design exotic devices such as acoustic super-lenses and cloaks. More recently, nonlinear AMM have been of interest as one means of creating non-reciprocal acoustic devices [Liang *et al*., Nat. Mater. **9** (2010)]. The efficiency of those types of non-reciprocal acoustic devices strongly depends on the material nonlinearity of one of its components and is therefore limited by existing fluids containing contrast agents. This work addresses this limitation by exploring the top-down design of a nonlinear AMM with non-resonant subwavelength structures that display non-monotonic pressure-volume strain response to generate an AMM with large parameters of nonlinearity. A simple one-way device consisting of a frequency-selective acoustic mirror and a nonlinear medium is described using a nonlinear multiscale model and a nonlinear propagation model that includes quadratic and cubic nonlinearity. Bayesian network classifiers (BNC) map regions of high performance at the device scale to each design level and intersect the high performance space across levels to identify multilevel solutions. Three design levels are considered: one-way performance, nonlinear acoustic propagation in the effective medium, and effective properties of the AMM based on subwavelength structure.

MONDAY MORNING, 2 NOVEMBER 2015 CITY TERRACE 7, 8:30 A.M. TO 11:40 A.M.

Session 1aSP

Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics: Direction of Arrival (DOA) Estimation, Source Localization, Classification and Tracking Using Small Aperture Arrays I

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

Geoffrey H. Goldman, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

Chair's Introduction—8:30

Invited Papers

8:35

1aSP1. Some aspects of signal and array processing with small aperture underwater acoustic arrays. Gerald L. D'Spain and Camille Pagniello (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

Signal and array processing with small aperture acoustic arrays is a critical issue when physical space for sensor deployment is limited, often the case in ocean acoustics. This presentation discusses processing issues and performance of various algorithms applied to data from these arrays. In 1954, Pritchard showed that as the ratio of the inter-hydrophone spacing to acoustic wavelength approached zero, the maximum directivity index approached a non-zero value. This phenomenon of "superdirectivity," where array weights become large and opposite in sign, can be understood in terms of a Taylor series expansion of the pressure field. This expansion also illustrates that an acoustic vector sensor is equivalent to a small volumetric hydrophone array, so that data processing issues for both array types is similar, although the primary sources of self noise differ. Formally, measurement of direction of arrival (DOA) is a parameter estimation problem. Therefore,

performance is quantified by the statistical distribution (bias and variance) of the estimates. Therefore, although the spatial resolution of adaptive beamformers and the array gain may be higher than conventional approaches, the statistical estimation performance may be similar. These points will be illustrated using actual ocean acoustic data. [Work supported by the Office of Naval Research.]

8:55

1aSP2. Symmetric small-aperture arrays for three-dimensional bearing estimation. David C. Swanson (Appl. Res. Lab, Penn State Univ., PO Box 30, State College, PA 16804, dcs5@arl.psu.edu) and Richard L. Culver (Penn State Univ., Chevy Chase, MD)

Small aperture arrays (size less than a wavelength) can be used for passive direction of arrival (DOA) estimation of both broadband and narrowband signals in the frequency domain. Phase differences across the array are measured in the frequency domain and can be spectrally averaged for stationary DOA and frequencies if desired. Data windowing will bias the DOA measurement toward the center of the FFT data buffer and is useful to prevent spectral leakage from strong target signals overshadowing weak target signals. The array is capable of measuring multiple target DOAs so long as each target produces unique frequencies. Broadband signals from targets can be collected from the FFT bins and grouped by arrival angle using a bearing histogram. The Cramer-Rao lower bound (CRLB) for bearing accuracy is presented as a function of frequency, aperture, and signal-to-noise ratio (SNR). The reduction in DOA accuracy due to a small aperture can be overcome if the SNR is sufficiently high. Using symmetry in 2D and 3D array geometries, we show how the azimuth and elevation angles can be separately measured. Examples are given for several single and multiple targets and different array shapes.

9:15

1aSP3. Direction-of-arrival estimations based on a two-microphone array using two levels of Bayesian inference. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu) and Jose Escolano (Sophandrey Res., Brooklyn, NY)

Due to a wide variety of potential applications, such as in video conferencing, mobile devices, and robotics, sound source localization using a small aperture consisting of only two microphones has been actively investigated. Based on the observed time-differences of arrival between sound signals, a probability distribution of the direction of the sources is derived to estimate the actual direction of sources. Many existing algorithms, however, assume a given number of sound sources. This paper describes a recent development in a model-based Bayesian probabilistic approach by Escolano *et al.* [J. Acoust. Am. Soc. 135, 742–751 (2014)], which allows both the number and direction of speech sources to be inferred. This paper will demonstrate that a unified framework encompassing two levels of Bayesian inference, model selection and parameter estimation, can be efficiently applied in this challenging task. This paper will also present different experimental setups and scenarios to investigate the performance of the proposed method.

Contributed Paper

9:35

1aSP4. Multiple and single snapshot compressive beamforming. Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu), Angeliki Xenaki (Dept. of Appl. Mathematics and Comput. Sci., Tech. Univ. of Denmark, Kgs.Lyngby, Denmark), and Christoph F. Mecklenbrauker (Christian Doppler Lab, Inst. of Telecommunications, TU Wien, Vienna, Austria)

For a sound field observed on a sensor array, compressive sensing (CS) reconstructs the direction-of-arrival (DOA) of multiple sources using a sparsity constraint. The DOA estimation is posed as an underdetermined problem by expressing the acoustic pressure at each sensor as a phase-lagged superposition of source amplitudes at all hypothetical DOAs. Regularizing with an ℓ_1-norm constraint renders the problem solvable with convex optimization, and promoting sparsity gives high-resolution DOA maps. Here, the sparse source distribution is derived using maximum a posteriori (MAP) estimates for both single and multiple snapshots. CS does not require inversion of the data covariance matrix and thus works well even for a single snapshot where it gives higher resolution than conventional beamforming. For multiple snapshots, CS outperforms conventional high-resolution methods, even with coherent arrivals and at low signal-to-noise ratio. The superior resolution of CS is demonstrated with vertical array data from the SWellEx96 experiment for coherent multi-paths.

Invited Papers

9:50

1aSP5. Small aperture microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoust., Burlington, VT)

Superdirectional microphone arrays are arrays that are typically smaller than the acoustic wavelength and attain directional gains that exceed those of a classical delay-sum beamformer. Early superdirectional microphones utilized a combination of a velocity microphone along with a pressure microphone. Later, a single sensor superdirectional microphone was discovered that utilized the combination of pressure and pressure-difference (which is proportional to the acoustic particle velocity) through appropriate porting of the incident acoustic field to both sides of the microphone diaphragm. In the early 1980s, with the advent of multichannel FFT analyzers, there was a renewed interest in using superdirectional microphone arrays to estimate the acoustic intensity and to calculate the acoustic power flow from sources. The estimation of the vector acoustic intensity led to many transducer designs that utilized up to 6 pressure microphones and commensurate signal processing. In this talk we will show some of the early acoustic intensity probes that were designed to estimate the acoustic intensity. These small arrays have been utilized in many other beamforming applications. We will show one where a superdirectional beamformer is also capable of source direction finding by utilizing some simple adaptive beamformer implementations.

10:10–10:25 Break

10:25

1aSP6. On robust time delay estimation in room acoustic environments. Jingdong Chen (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern Polytechnical Univ., 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org), Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, QC, Canada), and Gongping Huang (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern Polytechnical Univ., Xi'an, Shaanxi, China)

Time delay estimation, which serves as the first stage that feeds into subsequent processors for localizing and tracking radiating sources, has been an active research topic for decades. This paper deals with time delay estimation in room acoustic environments. The major focus is on developing algorithms that can achieve robust time delay estimates in reverberant and noisy environments. Three approaches are investigated, i.e., the blind channel identification based method, the multichannel cross correlation based technique, and a method based on the use of the Householder transformation. The first approach achieves its robustness with respect to reverberation by taking into account reverberation in the problem formulation; the second one improves the robustness by exploiting the redundancy provided by multiple microphones, and the third method attempts to separate noise and reverberation from the signal components. Experimental results are presented to illustrate their performance and pros and cons in room acoustic environments where reverberation and noise are commonly encountered.

10:45

1aSP7. Small infrasonic arrays for direction of arrival estimation. W. C. Kirkpatrick Alberts and Stephen M. Tenney (US Army Research Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net)

Long infrasonic wavelengths suggest a requirement for large arrays. This coupled with constraints in space can lead to array sizes and shapes that must strike a balance between the available resources and the tasks of the array; rarely are these arrays able to be installed in an optimal configuration. Further, typical frequencies of military interest span an order of magnitude in wavelength, roughly 17 to 170 m. However, standard beamforming techniques yield acceptable results even in frequency ranges where the spatial limitations of the available real estate force the size of the array to be much smaller than the wavelengths of a given signal of interest. Results acquired by processing several spatially constrained infrasonic arrays using standard techniques, e.g., minimum variance distortionless response (MVDR), will be discussed.

11:05

1aSP8. Comparison of direction on arrival estimates from a co-located vector sensor and horizontal line array. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington and Appl. Phys. Lab., Seattle, WA)

Direction of arrival (DOA) estimates made simultaneously by a vector sensor and a co-located 1.4 m long horizontal line array (HLA) are compared. The 8-channels of data, corresponding to four HLA pressure sensors and a vector sensor (3-orthogonal particle acceleration sensors + 1 pressure sensor), were coherently recorded by an autonomous bottom-deployed system. The system was deployed 4 km off the coast of Panama City, FL, in water 20 m deep. Recordings were made of a series of 1–4 kHz transmissions (100-ms duration) produced by source suspended 12 m below a drifting vessel. GPS measurements of the source location for each transmission show a source heading changing by 90 degrees relative to the sensor system as the source passes to within 200 m before opening to range 1200 m. The performance of DOA estimates from vector sensor measurements of active intensity and from HLA beamforming, using both conventional methods and compressive sensing, are assessed relative to the measured source bearing. At certain source locations DOA estimates deviate significantly from the source bearing. This effect is shown to be caused by destructively interfering modes, which can produce DOA estimates in the opposite direction of the source bearing.

Contributed Paper

11:25

1aSP9. Analysis and design of multi-pole vector sensor arrays. Junyuan Guo, Shie Yang, and Shengchun Piao (College of Underwater Acoust. Eng., Harbin Eng. Univ., NO.145, Nantong St., Nangang District, Harbin, Heilongjiang Province, Harbin 150001, China, guojunyuan@hrbeu.edu.cn)

Super-directive sensor arrays, due to their small size and enhanced directivity, are quite promising in signal-to-noise enhancement and target direction finding. An array design method, which can build a desired beam pattern using compact multi-pole sensor arrays, is presented in this paper. The proposed method extracts multi-pole modes of different order from the spatial differentials of the sound fields which are measured by a uniform square array composed with hydrophones and particle velocity sensors. It can achieve more benefits with the employment of particle velocity sensors because of their inherent dipole directivities. The spatial differential in the extraction procedure is approximated with spatial difference and the approximation decreases the beam pattern performance. The performance degradations in multi-pole beam patterns are investigated for different incident wave frequency. Simulation and analysis show that the array gain is to a certain extent robust to uncorrelated noise.

Session 1pAA

Architectural Acoustics: Sustainability and Acoustics

Lucky S. Tsaih, Cochair
Dept. of Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd., Sec. 4, Taipei 10607, Taiwan

Ronald Eligator, Cochair
Acoustic Distinctions, 145 Huguenot Street, New Rochelle, NY 10801

Chair's Introduction—2:00

Invited Papers

2:05

1pAA1. Synergies between high performance buildings and good acoustics. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Post-occupancy surveys and anecdotal evidence suggest that "green" buildings tend to have worse acoustics than new "non-green" and older buildings. While some aspects of green building design such as the increased use of glass, increased exposure of hard finishes, and the removal of barriers between work stations can create situations that reduce the acoustic performance of interior spaces, all is not lost. Many energy efficiency innovations can actually improve acoustic performance. In this presentation, some of the synergies between acoustics and high performance building design will be discussed.

2:25

1pAA2. A look at designing and integrating acoustics and sustainability. Shane J. Kanter, Robin Glosemeyer Petrone, Carl Giegold, Greg Miller, Scott Pfeiffer, and Brad Fritz (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

Designing with sensitivity toward our natural environment relates not only to the resources that are used in the design, construction, and ongoing operation of buildings, but also to creating environments inspired by nature to achieve a level of comfort in our surroundings. Green building design is by no means at odds with the goals of acoustic design. An overarching byproduct of good design should be an environment that is both pleasing to the ear and to the earth's resources. These principles can be subtle and subdued, such as in the reuse of building materials in the Writer's Theatre and the design of buildings to be flexible insuring longevity for Duke University. Or, rather than pouring energy into creating materials, buildings for music can be engineered with stiffness in mind to maintain energy and body, such as in the new orchestra shell at the Lyric Opera in Chicago. More traditionally, the Tower at PNC Plaza's tagline is the "world's greenest skyscraper;" yet, the interior of the office space shows minimal obvious signs of LEED influence. Alternatively, nature's touch is right on the sleeve of the National Research Defense Counsel project with the inclusion of foliage grown directly on the walls.

2:45

1pAA3. Understanding and communicating the importance of sustainable acoustics. David S. Woolworth and Amanda G. Higbie (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

A green or sustainable approach to acoustics design extends far beyond the use of green materials or meeting LEED requirements. This paper addresses the importance of understanding and using the language of sustainability, the gray area of acoustics and general exclusion from the (Mechanical-Electrical-Plumbing based) engineering conversation, and the benefits of integrated design and early inclusion of acoustics in the design process. Included will be tools and studies to communicate the advantage of sustainable acoustics practices to architects and owners.

3:05

1pAA4. Creating sustainable rooms through remediation. Joseph A. Keefe (Ostergaard Acoust. Assoc., 200 Executive Dr., STE 350, West Orange, NJ 07052, jkeefe@acousticalconsultant.com)

The author's consulting experience is that typical current sustainable design practices do not greatly affect acoustical design choices. However, many spaces are not "sustainable" due to poor acoustical performance (e.g., excessive reverberation). A few case studies of room investigations are presented, some with remediation efforts and some without, in order to evaluate how rooms can be made sustainable in the most important way: functional for their users.

3:25

1pAA5. The harmonization of sustainability and acoustics. Keely Siebein and Marylin Roa (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com)

A theoretical look will be presented at how building and environmental acoustics fall into the larger context of sustainability. The 2005 World Summit on Social Development identifies sustainable development concepts as being: Social, Environment, and Economic. Each concept is examined in terms of how it relates to acoustics. For example, Social relates to communication and human interaction. Whether it is optimizing speech communication in schools, or global networking with colleagues via Internet teleconferencing, or listening to music performances in high-fidelity acoustic environments; the idea of communication is at the root of the social fabric of our society. Environment relates to materials' complete lifecycle from raw material to disposal, using recyclable materials or renewable materials that are locally harvested, reducing unwanted sonic impacts into the surroundings, and creating pleasing sounds in the environment. Economic relates to 1st cost, lifecycle cost, embodied energy, and productivity of space and users in the space. Research has shown that acoustics impacts productivity in offices, healthcare occupancies, and schools. Finding ways to navigate between competing goals in terms of other building attributes is also discussed.

3:45

1pAA6. Integrating acoustics for sustainability. Ronald Eligator (Acoust. Distinctions, 145 Huguenot St., New Rochelle, NY 10801, religator@ad-ny.com)

Sustainability is frequently correlated with energy efficiency and environmentally sensitive design. Issues such as energy use, daylight harvesting, waste management, materials manufacturing, and recyclability are all considered. More difficult to quantify but of great importance in determining whether a project is truly sustainable is the degree to which a building or space appropriately serves the function for which it was designed. Thoughtful integration of acoustic design into building projects is required to ensure the finished product will meet the design goals of the client and users, including acoustic performance requirements. Without early integrated design efforts, the need for immediate renovation or other changes to a newly completed project due to acoustic deficiencies becomes much more likely. This presentation will provide case histories of projects made successful, and therefore sustainable, through an integrated design approach. Issues related to these successful outcomes will be discussed, including budget pressures, education of the client and design team regarding the effect design decisions have on the functionality and usability of spaces, and the importance of relationship building in encouraging design team members to take on design ideas which might challenge past approaches or assumptions.

Contributed Paper

4:05

1pAA7. The Lane Community College Alveolus: A case study of sound isolation in an educational building with naturally ventilated cooling. Kent McKelvie (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776, kmckelvie@cavtocci.com)

Providing adequate sound isolation in a building that incorporates naturally ventilated cooling presents many challenges and requires attention to detailing, along with close coordination between the acoustical consultant, architect, and project mechanical engineer. This presentation will walk through the design decisions and considerations for sound isolation at the Health and Wellness Facility for the Lane Community College in Eugene Oregon. The Lane Community College—Health and Wellness Facility is a LEED Gold certified education building that uses natural ventilation for building cooling. The building incorporates a 38-foot high by 180-foot long "lung", dubbed the Alveolus. This lung bisects the building along its full length and acts as a central chimney for evacuating warm air. Additionally, the "lung" space is constructed of translucent material so the large space can act as a light well. Using a clear (or semi-clear) polymer paneling system, the ventilation, natural lighting, and sound isolation needs were successfully balanced to meet the needs and desires of the Lane Community College.

Session 1pAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Comparative Neurophysiology of the Auditory System II: Session in Honor of Albert Feng

Andrea Simmons, Cochair
Brown University, Box 1821, Providence, RI 02912

Peter M. Narins, Cochair
Integrative Biology & Physiology, UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095

Invited Papers

1:30

1pAB1. Albert Feng and target ranging by echolocating bats. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

Ultrasonic biosonar sounds propagate out to a target and return as echoes. Simultaneously, phasic-on neural responses evoked by each outbound sound propagate from one subpopulation of neurons to another at different latencies in the inferior colliculus, creating delay-lines carrying a replica of the broadcast. When each echo returns, it evokes new on-responses that travel over similar subpopulations of neurons, but lagging behind responses to the broadcast according to the delay of the echo. Albert Feng's discovery of delay-tuned neurons in the bat's nucleus intercollicularis revealed that bats "read out" echo delay from the delay-lines by detecting the succession of coincidences between responses to the echo and to the emission. Subsequent work in multiple labs demonstrated the ubiquity of delay-tuning in bats. Working in the inferior colliculus, Al Feng showed how the delay-line responses are restricted to single spikes by initial inhibition that determines each neuron's characteristic on-response latency followed by new inhibition that suppresses any subsequent multiple spikes. The interplay between timed and triggered inhibition defines the delay-lines, while delay-tuned coincidence-detecting neurons from inferior colliculus to superior colliculus guide the bat and from inferior colliculus to auditory cortex create the bat's perceived images. [Work supported by ONR.]

1:50

1pAB2. Strategies for an echolocating FM bat, *Pipistrellus abramus*, to listen to weak echoes. Hiroshi Riquimaroux (Shandong Univ.-Virginia Tech Joint Program, Shandong Univ., 27 Shanda Nanlu, Jinan, Shandong 250100, China, hiroshi_riquimaroux@brown.edu)

In Japanese house bat, *Pipistrellus abramus*, a typical FM-bat, neurophysiological investigation in the inferior colliculus showed that about half of neurons was tuned to the terminal frequency of the downward FM seep or pseudo-CF frequency for searching insects, which was around 40 kHz. Their audiograms determined by the auditory brainstem response (ABR) and the compound action potentials (CAP) of the cochlear nerve also show the lowest threshold or maximum amplitude to be found at around 40 kHz. However, we could not find a sharp notch in their audiogram around 40 kHz, which we have usually found in CF-FM bats at round their reference CF frequency. We examined the peripheral system of Japanese house bat how they extract signals of a flying insect. A particular attention was paid on their terminal frequency of FM sweep or pseudo-CF frequency. [Research supported by MEXT of Japan.]

2:10

1pAB3. Three-dimensional space representation by echolocation in bats. Melville J. Wohlgemuth, Ninad B. Kothari (Psychol. and Brain Sciences, Johns Hopkins Univ., Baltimore, MD), and Cynthia F. Moss (Psychol. and Brain Sciences, Johns Hopkins Univ., Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

Echolocating bats are equipped with a biological sonar system that permits spatial navigation and target tracking in complete darkness. By actively controlling the directional aim, timing, frequency content, and duration of echolocation signals to "illuminate" the environment, the bat directly influences the acoustic input available to its sonar imaging system. Detailed analyses of the bat's sonar behavior suggest that the animal's actions play into a rich 3-D representation of the environment, which then guides motor commands for subsequent call production, head aim, and flight control in an adaptive feedback system. Studies of the bat's sonar behavior have motivated neurophysiological studies of the midbrain superior colliculus (SC), a structure implicated in sensorimotor integration and spatial orientation. Using multichannel silicon probe recordings from the freely echolocating big brown bat, we characterized response profiles of single neurons, as the animal tracked moving targets from a stationary position on a platform and in free flight. Our data show premotor activity prior to sonar vocalizations and responses to echoes from objects. Echo responses depended on both the azimuth and delay of sonar returns. These findings demonstrate both auditory and premotor specializations in the bat SC, which contribute to the animal's 3-D representation of space.

2:30

1pAB4. Neural processing of novel sounds in the rat's inferior colliculus. Huiming Zhang, Chirag Patel, and Ariana Lumani (Dept. of Biological Sci., Univ. of Windsor, 401 Sunset Ave., Windsor, ON N9B3P4, Canada, hzhang@uwindsor.ca)

A novel sound is an occasionally occurring acoustic event in a constant/repetitive acoustic environment. Auditory neurons display sensitivity to a novel sound by reducing action potential discharges over repetitive acoustic stimulation and restoring discharges upon presence of a novel sound. We used the rat as an animal model and conducted *in vivo* neurophysiological recordings to examine how novel sounds are processed in the midbrain auditory structure, the inferior colliculus (IC). The IC receives convergent inputs from structures including those with neurons sensitive to novel sounds. These inputs have diverse sensitivities to temporal, spectral, and directional acoustic cues and produce excitatory/inhibitory synaptic events with different time courses in IC neurons. Integration among these inputs in IC neurons provides a likely physiological basis for processing the novelty of a sound and other dynamic acoustic characteristics. We used both closed- and free-field stimuli to examine responses to oddball paradigms. Our results indicate that neurons sensitive to novel sounds existed in the IC. These neurons typically have onset firing patterns; and their sensitivity to a novel sound is affected by the spatial relationship between the novel sound and a repetitive standard sound in the environment. [Research supported by NSERC of Canada.]

2:50–3:05 Break

3:05

1pAB5. Sound localization in the frog after Albert Feng: Filling in the blanks. Peter M. Narins (Integrative Biology & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095, pnarins@ucla.edu)

In his Ph.D. thesis, Albert described his elegant behavioral study demonstrating that two ears are necessary for frogs to localize sound. Next, he elucidated a series of mechanisms in the frog CNS that is responsible for encoding sound source direction. Inspired by this work, we examined some of the peripheral mechanisms involved in sound processing in the frog. For example, frogs and toads are capable of producing calls at potentially damaging levels that exceed 110 dB SPL at 50 cm. In most frog species, the tympanic membranes (TMs) communicate directly via the large, permanently open Eustachian tubes, resulting in an inherently directional asymmetrical pressure-gradient receiver. One active mechanism for auditory sensitivity reduction involves the pressure increase during vocalization that distends the TM, reducing its airborne sound sensitivity. Another states that if sounds generated by the vocal folds arrive at both surfaces of the TM with nearly equal amplitudes and phases, the net motion of the eardrum would be greatly attenuated. Both mechanisms can explain the results of our direct TM measurements. Moreover, we have measured this eardrum motion under acoustic clamp conditions and shown that blocking the internal interaural connection results in nearly azimuth-independent eardrum velocity, consistent with Albert's results.

3:25

1pAB6. Sound-triggered suppression of neuronal firing in the auditory cortex: Implications to the residual inhibition of tinnitus. Alexander Galazyuk (Anatomy and Neurobiology, Northeast Ohio Medical Univ., 4209 St. Rt. 44, Rootstown, OH 44272, agalaz@neomed.edu)

Tinnitus can be suppressed briefly following the offset of an external sound. This phenomenon, termed "residual inhibition," has been known for almost four decades, although its underlying cellular mechanism remains unknown. In our previous work, we have shown that the majority of neurons in the inferior colliculus (IC) exhibit long lasting suppression of spontaneous activity following the offset of an external sound. The time course of suppression corresponded to the time course of residual inhibition in tinnitus patients. If the suppression is an underlying mechanism, the auditory cortex (AC) neurons should also exhibit suppression because residual inhibition of tinnitus is a perceptual phenomenon. To test this hypothesis, we studied sound evoked suppression in AC neurons of awake CBA/CaJ mice using extracellular recording. Pure tones at neurons' characteristic frequency and/or wideband noise stimuli 30 s duration were delivered in the free-field. We found that AC neurons exhibited sound-triggered suppression of their spontaneous firing. Similar to the IC, the duration of this suppression was roughly corresponded to the stimulus duration (about 30 s). AC neurons also showed longer suppression to tones at their characteristic frequency than to wideband noise. Our data suggest that suppression may be a neural correlate of the residual inhibition of tinnitus in humans. [Work supported by Grant R01 DC011330 from the National Institute on Deafness and Other Communication Disorders.]

3:45

1pAB7. fMRI and electrophysiology of cortical layer-dependent processes in primary sensory cortex. Jozien Goense (Inst. for Neurosci. & Psych., Univ. of Glasgow, 58 Hillhead St., Glasgow G12 8QQ, United Kingdom, jozien.goense@glasgow.ac.uk)

To understand cortical function, it is important to better understand the cortical functional units, i.e., its columns and layers. High-resolution fMRI can offer tremendous advantages for the study of cortical circuits *in vivo*, evidenced by the increasing interest in high-resolution fMRI. However, many questions remain about the size of the cortical features that can be resolved, and how neural activity gives rise to the blood oxygenation dependent (BOLD) signal measured with fMRI. We use high-resolution fMRI combined with electrophysiology to study laminar processing, with the aim of gaining insight into the layer-dependent neural processing and the mechanisms of neurovascular coupling in the primary visual cortex (V1) and temporal lobe of awake and anesthetized macaques. We investigated whether laminar differences in the BOLD, cerebral blood flow (CBF), and volume (CBV) responses can be detected for excitatory and inhibitory stimuli, and found that the mechanisms for positive and negative BOLD responses differ, but also that neurovascular coupling differs in the cortical layers. Furthermore, neuromodulators such as dopamine can alter neurovascular coupling. Our results suggest that neurovascular coupling depends on multiple factors, and that the combination of high-resolution fMRI with electrophysiology can be used to resolve neurovascular coupling in functional microcircuits.

Session 1pAO

Acoustical Oceanography, Signal Processing in Acoustics, Underwater Acoustics, and Animal Bioacoustics: Acoustics of High Latitude Oceans II

Aaron Thode, Cochair
SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Contributed Papers

1:15

1pAO1. Development of acoustic remote sensing techniques for sea ice, oil under sea ice, and oil encapsulated in sea ice. Christopher Bassett, Andone C. Lavery, Ted Maksym (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, cbassett@whoi.edu), Jeremy Wilkinson (Br. Antarctic Survey, Cambridge, United Kingdom), Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Scott Pegau (Oil Spill Recovery Inst., Anchorage, AK)

Recent decreases in sea ice cover have provided new opportunities for the shipping industry and stimulated further interest in hydrocarbon extraction in Arctic waters, thereby also increasing the risk of an oil spill in ice covered waters. To support oil spill response there is a need to develop practical remote sensing techniques to detect, quantify, and map oil that is under or encapsulated in sea ice. To address this need, a series of experiments were conducted at the Cold Regions Research and Environmental Laboratory (CRREL, Hannover, NH). During these experiments, different amounts of crude oil were injected underneath artificially grown sea ice of different thicknesses. The ice and oil were monitored by a suite of instruments located in the water column, including cameras, laser fluorometers, and multibeam, narrowband, and broadband acoustic backscattering systems. In addition, temperature and salinity profiles were conducted routinely, and ice cores were collected and imaged using a micro-CT scanner. Results from the broadband acoustic backscattering system are presented, and the relative merits of this approach for the remote detection and quantification of oil under and in sea ice are discussed. In addition, the utility of the acoustic systems for studies of ice physics are also discussed.

1:30

1pAO2. A seven-year review of ambient acoustic environments in the Beaufort Sea. Kerri D. Seger, Aaron M. Thode (Scripps Inst. of Oceanogr., 9331 Discovery Way, Apt C, La Jolla, CA 92037, kseger@ucsd.edu), Susanna B. Blackwell, and Katherine H. Kim (Greeneridge Sci., Inc., Santa Barbara, CA)

Over seven seasons (2007 to 2014) Greeneridge Sciences, Inc., deployed passive acoustic recorders (DASARs) between August and October at five sites in the Beaufort Sea off the Alaskan North Slope to collect acoustic data during the fall bowhead whale migration. Each site consisted of 7–11 DASARs, arranged in triangular grids with 7 km spacing between each unit. The shallowest DASAR unit in each set was deployed 15–33 km due north of the coast in 22–39 m of water. The acoustic environments between sites differ due to varying ocean depths and sound source contributions (whales, winds, and human activities). Here, we conduct a multi-year bulk analysis of the summer Beaufort Sea acoustic environment, comparing noise properties as a function of both distance offshore and longitude within a given year, and across sites across all eight seasons. Comparisons were conducted by investigating the percentile distributions of ambient noise intensity over several bandwidths that are representative of the different sound sources present in the Beaufort Sea. Results suggest that variations in the ambient noise field across sites and years create spatially heterogeneous acoustic environments that must be accounted for when addressing responses of bowhead whales to industrial noise.

1:45

1pAO3. Modal analysis of broadband signal arrival on Beaufort shelf. Mohsen Badiey, Lin Wan, Andreas Muenchow (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), and David Knobles (Appl. Res. Lab., Univ. of Texas, Austin, TX)

Using previously reported temperature and salinity data collected over the past decade, time evolving sound speed profiles were constructed and used (Prog. Oceanography, 127, pp.1-20, 2014) as input to a parabolic equation model. Modeling of broadband acoustic signal propagation in the Arctic shelf-basin region during ice free sea surface season was reported recently (J. Acoust. Soc. Am. **136**(4), Pt. 2, 2317, 2014). It was shown that when broadband acoustic signals propagate from deep water to shallow water, the modal dispersion is changed. The change is mostly affected by the water thermocline behavior in time and space and by source depth. The modal arrival structure can be influenced by a number of factors including the degree of Pacific warm water intrusion, upwelling of the warm saline Atlantic water into the cold surface layer, the range-dependence of the water layers over the shelf break, and the depth of the source. In this paper, we analyze the range-dependent propagation in terms of adiabatic and coupled mode regimes in order to explain the model results. [Work supported by ONR.]

2:00

1pAO4. Investigating the effects of ocean layering and sea ice cover on acoustic propagation in the Beaufort Sea. Jason D. Sagers, Megan S. Ballard, David P. Knobles (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu), Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Lewes, TX), and Andreas Muenchow (College of Earth, Ocean, and Environment, Univ. of Delaware, Lewes, DE)

Over the past several decades, the Beaufort Gyre has experienced changes in sea-ice freshwater accumulation and ocean stratification which has implications for long-range acoustic propagation. In this talk, acoustic propagation from the Canadian Basin to the Alaskan Beaufort Shelf is modeled using measurements of physical oceanography and sea ice. Water masses that impact acoustic propagation and stratification in the basin include the warm, saline Atlantic Water (AW), which is overlain by cooler, less-saline Pacific Winter Water (PWW). Oceanographic observations reveal intrusions of a warmer, fresher water mass called Pacific Summer Water (PSW). This water mass resides below the surface mixed layer, but

above the PWW and reduces acoustic interaction with the sea-ice canopy for source depths located in the halocline duct. Oceanographic data indicate that on the continental shelf, the PSW intrusion can be absent in the ice-covered months resulting in an upward refracting sound speed profile. Using measurements from ice-tethered profilers in the basin and oceanographic moorings on the shelf, we model the temporal and spatial variability of the acoustic field. The effect of scattering from the ice cover is included, with consideration given to the seasonal variability of sea-ice concentration, thickness, and acoustic properties. [Work supported by ONR.]

2:15

1pAO5. Laboratory measurements of high-frequency broadband acoustic backscatter from sea ice, oil under sea ice, and oil encapsulated in sea ice. Christopher Bassett and Andone C. Lavery (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, cbassett@whoi.edu)

To investigate the potential for detection of crude oil under sea ice using active acoustics techniques, measurements of high-frequency broadband backscattering (75–590 kHz) from crude oil of different thicknesses (0.7–8 cm) under, and frozen within, laboratory sea ice have been performed at the Cold Regions Research and Environmental Laboratory (Hannover, NH). Backscattering measurements were performed at normal and 20 degrees from normal incidence. The data have been analyzed in both the temporal domain and in the frequency domain, allowing scattered spectra from the oil and ice to be measured. The results show structure consistent with scattering from multiple interfaces following the injection of oil under the ice and during the subsequent encapsulation the oil layer. The acoustic estimates of oil thickness are in general agreement with ancillary measurements. The sound speed of the crude oil was separately measured over a range of relevant temperatures, both to inform a scattering model and to accurately infer the oil thickness. Predictions based on a simple scattering model for the frequency-dependent reflection coefficient of oil under ice agree well with the normal incidence measurements prior to oil encapsulation. At angles off normal incidence, volume inhomogeneities appear to dominate the scattering.

2:30

1pAO6. Passive acoustic monitoring and ambient noise in the high Arctic: Resolute Bay, Nunavut. Caitlin O'Neill (School of Earth and Ocean Sci., Univ. of Victoria, 304-1000 McKenzie Ave., Victoria, BC V8X 4C8, Canada, caitlin.v.oneill@gmail.com) and Svein Vagle (Fisheries and Oceans Canada, Victoria, BC, Canada)

Resolute Bay, a remote bay in the Canadian High Arctic to the north of Parry Channel, hosts diverse populations of marine mammals that migrate through the bay each year following food availability and/or oceanographic conditions. The changing climate combined with increasing anthropogenic activity in the Arctic make it important to create an ecosystem baseline from which to predict, understand, and monitor future changes. Passive underwater acoustic observations provide a non-invasive way to monitor marine mammal presence. Broadband noise (10 Hz to 48 kHz) was recorded by an Autonomous Marine Acoustic Recorder (AMAR) and marine mammal click detections were logged by two CPODs over a 5 month period from August to December 2013. Acoustic data were processed with click and tonal call detectors to determine marine mammal presence. Resolute Bay is ice-covered 10 months a year, leading to increased broadband ambient noise levels due to ice movement. During the short open-water period, vessel activity is common in the bay. The aim is to compare these two different ambient noise regimes and how they affect the effectiveness of passive acoustic marine mammal detections and tracking.

2:45

1pAO7. Observations of thermohaline sound speed structure in the Beaufort Sea in the summer of 2015. Dominic DiMaggio, Annalise Pearson, and John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, dfdimagg@nps.edu)

Moored observations of temperature and salinity were made in the depth of range 50–510 m between July 15 and August 15 2015 in the central Beaufort Sea, as part of the Canada Basin Acoustic Propagation Experiment (CANAPE) pilot study. This talk will present an analysis of the observed sound speed structure in terms of the fresh upper layer, the Pacific layers, and the Atlantic layer. The talk will also address the space/time scales of fluctuations associated with internal waves, eddies, and diffusive layering (stair cases).

3:00–3:15 Break

3:15

1pAO8. Bowhead whale localization using asynchronous hydrophones in the Chukchi Sea. Graham A. Warner, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper estimates bowhead whale locations and uncertainties from Bayesian inversion of modally dispersed calls recorded on asynchronous recorders in the Chukchi Sea, Alaska. Bowhead calls were recorded on a cluster of seven asynchronous ocean-bottom hydrophones that were separated by 0.5–7.5 km. A warping time-frequency analysis is used to extract relative mode arrival times as a function of frequency for nine frequency-modulated whale calls that dispersed in the shallow water environment. Each call was recorded on multiple hydrophones and the mode arrival times are inverted for: the whale location in the horizontal plane, source instantaneous frequency (IF), water sound-speed profile, subbottom layering and geoacoustic parameters, relative recorder clock drifts, and residual error standard deviation, all with estimated uncertainties. A simulation study shows that accurate prior environmental knowledge is not required for accurate localization. Joint inversion of multiple recorded calls is shown to substantially reduce localization, source IF, and relative clock drift uncertainties. Whale location uncertainties are estimated between 30 and 160 m and clock drift uncertainty is estimated between 3 and 26 ms. The clock synchronization provided by the inversion is sufficient for localizing other types of marine mammal calls using simpler time-difference-of-arrival methods.

3:30

1pAO9. Resolution, identification, and stability of broadband acoustic arrivals in Fram Strait. Matthew A. Dzieciuch, Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, mdzieciuch@ucsd.edu), Hanne Sagen, Stein Sandven, Florian Geyer, Mohamed Babiker (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Agnieszka Beszczynska-Möller (Inst. of Oceanology of the Polish Acad. of Sci., Sopot, Poland), and Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Fram Strait is the only deep-water connection between the Arctic and the world oceans. An acoustic system for tomography, glider navigation, and passive listening was installed in the central, deep-water part of the Strait during 2010–2012, with the primary objective of improving the estimates of transport through the Strait. Previous tomographic measurements have relied on the travel times of resolved, identified, and stable acoustic arrivals. The oceanographic conditions and highly variable sea ice in Fram Strait provide an acoustic environment that differs substantially from those in other tomographic experiments, however, and results in complex arrival patterns. Comparisons of the measured arrival patterns with predictions based on hydrographic sections show that it is difficult to resolve and identify individual arrivals in the early part of the arrival patterns. In addition, the early arrivals are unstable, with the arrival structure changing significantly over time. Later arrivals that are surface-reflected, bottom-reflected tend to be easier to resolve and identify, as well as more stable. The implication is that inverse methods need to use fluctuations in the overall structure of the early arrivals, which tend to sample the ocean similarly, in combination with the travel times of the later arrivals.

3:45

1pAO10. Modeling high-frequency acoustic backscatter for remote sensing of oil under sea ice and oil encapsulated in sea ice. Dajun Tang, Derrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105, djtang@apl.washington.edu), Christopher Bassett, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

High-frequency acoustic systems provide one of the potentially practical means for remote sensing of oil-in-ice. Sea ice is a complex medium that reflects, refracts, and scatters sound waves in complicated manner. The presence of oil adds additional complexity. A set of high-frequency acoustic data were taken at the Cold Regions Research and Environmental Laboratory (Hannover, NH) where crude oil was injected underneath artificial sea ice. A physics-based model is developed to interpret the data. The model consists of four layers to respectively describe the homogeneous half-space of water, the heterogeneous oil layer under ice, a thin skeletal ice layer, and a thick layer for the body of ice. The skeletal layer is allowed to have very different properties from the body of ice. All the interfaces at the boundaries of the layers are assumed rough. The model predicts scattered sound intensity in the time domain. Model results are discussed when it is applied to the acoustics data taken at both normal and 20-degree oblique incidence angles. The utility and the applicability of the model to actual arctic environments are anticipated.

4:00

1pAO11. Modeling thermal fracturing of sea ice, a historical view. Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pstein@scisol.com)

Thermal fracturing of sea ice is an important mechanism not only for ambient noise generation, but also for climate change studies. It is possible that thermal fracturing is a fundamental mechanism behind the weakening of first and multiyear ice. During the late 1980s and early 1990s, a significant effort was conducted to model and measure the thermal fracturing of sea ice by the author and Dr. James K. Lewis. Here, we take a look back at that effort in the broader context of Arctic Ocean ambient noise and climate change.

4:15

1pAO12. Ambient noise in the Arctic Ocean measured with a drifting vertical line array. Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), and John A. Colosi (Naval Postgrad. School, Monterey, CA)

In mid-April 2013, a Distributed Vertical Line Array (DVLA) with 22 hydrophone modules over a 600-m aperture immediately below the subsurface float was moored near the North Pole. The mooring parted just above the anchor shortly after deployment and subsequently drifted slowly south toward Fram Strait until it was recovered in mid-September 2013. The DVLA recorded low-frequency ambient noise (1953.125 samples per second) for 108 minutes six days per week. Previously reported noise levels in the Arctic are highly variable, with periods of low noise when the wind is low and the ice is stable and periods of high noise associated with pressure ridging. The median noise level at 98 m depth during the first two weeks of May not far from the North Pole had a maximum between 10 and 20 Hz of approximately 75 dB re 1 μPa^2/Hz. The background noise levels are at times extraordinarily low, with the 10th percentile over a 108-minute period limited by the self-noise of the hydrophones above approximately 100 Hz (35 and 26 dB re 1 μPa^2/Hz at 100 and 1000 Hz, respectively). During this time, the median ambient noise levels increased with depth by roughly 3 dB between 100 and 600 m.

4:30

1pAO13. Mid-frequency attenuation estimates from deep-water experiments using eigenrays. Jit Sarkar, Christopher M. Verlinden, Jefferey D. Tippmann, William S. Hodgkiss, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92093-0238, bsarkar@ucsd.edu)

A collection of deep-water experiments were recently performed using a short vertical array cut for 7.5 kHz and a source transmitting tonals as well as chirps. Eigenray arrivals out to the first convergence zone were identified through matching the processed data to ray-tracing models, using environmental parameters measured during the experiments. The ray path and transmission loss information were used to produce an ocean-average attenuation result, and compared to decades-old attenuation models. Attempts were made at producing a layered estimate of attenuation using the experimental-simulation ray-matching — a technique previously proposed for inverting for ocean acidity. We present our ocean-average mid-frequency attenuation estimates, and our initial results for a layered attenuation inversion.

4:45

1pAO14. Long-term measurements of the directionality and active intensity of the underwater noise field in the shallow Beaufort Sea. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna Blackwell (Greeneridge Sci., Santa Barbara, CA), Kerri Seger (SIO, UCSD, La Jolla, CA), and Katherine Kim (Greeneridge Sci., Santa Barbara, CA)

In each of the past seven years, at least 35 Directional Autonomous Seafloor Acoustic Recorders (DASARs) have been deployed over a 280 km swath of the Beaufort Sea continental shelf (20–55 m depth) during the open-water season to monitor the westward bowhead whale migration. DASARs have one omnidirectional pressure sensor and two orthogonal particle velocity sensors that permit measurements of the azimuths of both transient and continuous sounds, including diffuse ocean noise. Here, we map the azimuthal directionality of the Beaufort ambient noise field as a function of frequency and location across all seven seasons. Dominant directionalities exist in the diffuse ambient noise field, which change with frequency, time, and location. We examine how localized storms, heavy whale calling activity, seismic exploration, and other industrial activities influence the noise directionality. We also examine how both the active and reactive intensity of the noise evolves with frequency and time, by comparing the phase relationships between pressure and particle velocity. The directionality and active intensity help identify source mechanisms and help determine whether long-term changes in the ambient noise environment are occurring. [Work sponsored by the Shell Exploration and Production Company.]

Session 1pBA

Biomedical Acoustics: Medical Ultrasound and Imaging

Siddartha Sikdar, Chair
Bioengineering, George Mason University, 4400 University Drive, MS 2A1, Fairfax, VA 22030

Contributed Papers

1:30

1pBA1. High-frequency ultrasound of histology mimicking phantoms for evaluating breast cancer surgical margins. Nicole Cowan (Biotechnology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, ncowan18@gmail.com), Zachary A. Coffman (Biology, Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The ability to differentiate between malignant and normal tissues in surgical margins during breast cancer surgery would reduce the risk of local recurrence and subsequent surgeries. Clinical studies at the Huntsman Cancer Institute show that high-frequency (HF) ultrasound (20–80 MHz), and the parameters peak density (number of spectral peaks and valleys in the 20–80 MHz range) and attenuation, are sensitive to breast tissue pathology. The objective of this study was to determine the effect of tissue microstructure on these parameters using histology mimicking phantoms. Phantoms were created from distilled water, agarose powder, 10X TBE stock solution, and polyethylene microspheres to simulate breast tissue histology. Microsphere size (59–925μm diameter) and weight percent (0.00–0.06g) were varied in the experiments. Pitch-catch measurements were acquired using 50-MHz transducers, a HF pulser-receiver, a 1-GHz digital oscilloscope, and glycerol as the coupling agent. Both peak density and attenuation showed sensitivity to microsphere diameter and the number of scatterers present. Peak density followed an inverse-size relationship to microsphere diameter, whereas attenuation showed a sensitivity to the total weight percent of scatterers. The phantom results confirm that peak density and attenuation are complementary parameters for characterizing breast tissue pathology and validate the clinical studies.

1:45

1pBA2. Measuring the cytoskeletal properties of cell cultures using high-frequency ultrasound. Ashley Behan (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, ashleyrosales92@gmail.com), Caitlin Carter (Biotechnology, Utah Valley Univ., Orem, UT), Amy A. LaFond, Dolly A. Sanjinez, Mandy H. Marvel (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency ultrasound (10–100 MHz) has been demonstrated to be sensitive to cell cytoskeletal changes. Cytoskeletal properties determine the biomechanical characteristics of cells and their role in many biomolecular processes. Examples include the aggressiveness and metastatic potential of breast cancer subtypes, T-cell activation during immune responses, and microtubule disintegration in Alzheimer's disease. The objectives of this work were to optimize the use of high-frequency ultrasound to subtype breast cancer cells and to acoustically measure cytoskeletal modifications. Pulse-echo measurements of 7 breast cancer cell lines of different molecular subtypes were acquired over a 2.5-year period using a 50-MHz transducer immersed in the growth media of monolayer cell cultures. Cell reflections were isolated from the interfering cell-culture plate reflections, spectrally analyzed using Gaussian curve fits, and spectrally classified using a heat map. The heat map displayed distinct patterns that differentiated the cell lines by molecular subtype. Cell cultures were also treated with colchicine and sphingosylphosphorylcholine to observe modulation of the microtubule and actin components. Cell waveforms and spectra displayed time-dependent changes due to chemical modification of the cytoskeleton. These results further verify and improve the noninvasive use of high-frequency ultrasound to differentiate breast cancer subtypes and to monitor cytoskeletal alterations in real time.

2:00

1pBA3. High-frequency ultrasound for rapid detection of skin cancer and other pathologies: Studies on porcine tissue. Benjamin F. Finch, James P. Pacheco (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, benjaminfinchmed@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The Breast Cancer Research Laboratory at Utah Valley University has been developing high-frequency (HF) ultrasound (20–80 MHz) to differentiate between malignant and benign tissues. Results from our breast cancer clinical trials thus far are promising, with high sensitivities and specificities. The objective of this study was to determine whether HF ultrasound can provide pathology sensitive measurements for diagnosing skin cancer and distinguishing between tissue structures. Formalin-preserved porcine tissues were first used to test the feasibility of the approach. The results show that both spectral peak density and wave velocity were sensitive to structure, and that normal skin tissue was significantly discernible from other tissues. An 80-patient clinical study is currently being conducted at the Huntsman Cancer Institute with the collection of ultrasonic measurements from at least 320 skin biopsies. Multiple pulse-echo and through-transmission measurements are acquired from each biopsy specimen. The ultrasound data are correlated to conventional pathology to determine sensitivity and specificity. If successful, HF ultrasound may provide an earlier diagnosis of skin cancer and a rapid, intraoperative method for differentiating between melanoma and benign pathologies. By improving on current methods, HF ultrasound may provide dermatologists with faster and more accurate results, and thus better patient treatment and outcomes.

2:15

1pBA4. High-frequency ultrasound (20–80 MHz) for analyzing breast cancer surgical margins: A 73-patient clinical study. Amy A. LaFond (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, fairbrother.aa@gmail.com), Caitlin Carter (Biotechnology, Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), Rachel E. Factor (Pathology, Univ. of Utah, Salt Lake City, UT), Leigh A. Neumayer (Surgery, Univ. of Arizona, Tucson, AZ), and Timothy E. Doyle (Physics, Utah Valley Univ., Orem, UT)

Results from a 2010 pilot study indicate that the peak densities of high-frequency (HF) ultrasonic spectra (20–80 MHz) correlate to a wide range of margin pathologies from breast conservation surgery (BCS). Utah Valley University and the Huntsman Cancer Institute conducted a follow-up study to determine the sensitivity and specificity of HF ultrasound for differentiating malignant from nonmalignant tissue in BCS margins. A 73-patient blind study was performed with conventional pathology used as the gold standard.

A total of 492 specimens were ultrasonically tested *ex vivo* and then sent to pathology for analysis. The margins were approximately 3x20x20 mm, with each sampled at 2–5 locations. The data were analyzed for malignancy using peak density. Results from the current study indicate that peak density can differentiate malignant from nonmalignant pathologies with an accuracy of 73.8%. Trends from the pilot study closely resemble this study's results. Application of these trends to the current study predicts that a multivariate analysis will yield much higher accuracy (84.1%), specificity (85.2%), and sensitivity (77.6%) values. The results show that HF ultrasound can provide rapid, intraoperative evaluation of surgical margins, thereby increasing the quality and efficacy of breast cancer surgery. [Funding provided by the Elsa U. Pardee Foundation.]

2:30

1pBA5. High-frequency ultrasound for evaluating margins during breast conservation surgery: Results from a 17-patient pilot study. Robyn K. Omer (Botany, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, robynkiraomer@gmail.com), Kristina M. Sorensen (Mathematics, Pennsylvania State College, State College, PA), Leigh A. Neumayer (Surgery, Univ. of Arizona, Tucson, AZ), Rachel E. Factor (Pathology, Univ. of Utah, Salt Lake City, UT), and Timothy E. Doyle (Physics, Utah Valley Univ., Orem, UT)

Obtaining negative (cancer-free) margins in breast conservation surgery (BCS) is essential for ensuring all of the cancer has been removed from the excision site. Several rapid, noninvasive cancer detection methods are therefore being investigated for the intraoperative evaluation of margin status. This study investigated high-frequency (HF) ultrasound (20–80 MHz) as an intraoperative margin evaluation technique during BCS. In a 17-patient pilot study at the Huntsman Cancer Institute, Salt Lake City, Utah, through-transmission and pulse-echo measurements were acquired from 53 positions on specimens including margins, tumors, lymph nodes, and fibroadenomas. Measurements were acquired with the use of two 50-MHz transducers, a HF square-wave pulser/receiver, a 500-MHz digital oscilloscope, and a notebook PC. Parameters calculated from the data included peak density (the number of peaks and valleys across the ultrasonic spectrum), attenuation, and the slope of the second Fourier transform. Statistical analysis of the data revealed that a multivariate analysis combining peak density and attenuation provided the highest accuracy and sensitivity for differentiating malignant from nonmalignant tissue. The multivariate analysis showed 81.1% accuracy, 76.9% sensitivity, and 85.2% specificity. The results demonstrate that HF ultrasound is competitive with 2D specimen mammography and radiofrequency spectroscopy for margin evaluations. [Funding provided by Utah Valley University.]

2:45

1pBA6. A compensation method of frequency-dependent attenuation for pulsed Doppler systems by adapting the transmitting waveform. Jun Nishimura, Yu Teshima, Shizuko Hiryu, and Iwaki Akiyama (Life and Medical Sci., Doshisha Univ., 202-5 Kouridzuka, Kodo, Kyotanabe, Kyoto 6100310, Japan, dmo5001@mail4.doshisha.ac.jp)

Pulsed Doppler systems require a high signal-to-noise ratio (SNR) to differentiate the low amplitude of the blood flow echoes from the floor noise. Conventional quadrature demodulation (QDM) assumes that the central frequency of the emitted pulse is the same as the received echoes and uses this frequency as demodulation frequency. Nevertheless, the central frequency of the received echoes is actually downshifted due to the frequency-dependent attenuation (FRDA). This downshifting produces loss of SNR due to discarding energy of the down-mixed signal. In this study, by estimating the spectrum of a Gaussian modulated pulse, the downshift caused by the FRDA was obtained to create a new compensated pulse with the aim of forcing the echo central frequency to match with the demodulation frequency. The method was evaluated by using Field II simulation. Considering an attenuation of 0.5 dB/MHz/cm, a 5 MHz transducer and a relative bandwidth of approximately 50%, at a depth of 16 cm, compensated pulses reduced the frequency shift from 1.11 to 0.32 MHz and the SNR degradation from 9.25 to 0.41 dB. [This study was supported by the MEXT-Support Program for the Strategic Research Foundation at Private Universities, 2013–2017].

3:00

1pBA7. The effect of ambient pressure on the color Doppler ultrasound twinkling artifact. Julianna C. Simon, Bryan W. Cunitz (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ. Moscow, Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), Wayne Kreider (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Jeffrey Thiel (Dept. of Radiology, Univ. of Washington Med. Ctr., Seattle, WA), James R. Holm (Ctr. for Hyperbaric Med., Virginia Mason Med. Ctr., Seattle, WA), Mathew D. Sorensen (Div. of Urology, Dept. of Veteran Affairs Med. Ctr., Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Recently, our group discovered that overpressure suppressed the color Doppler ultrasound twinkling artifact on *ex vivo* calcium oxalate monohydrate (COM) kidney stones, suggesting that trapped microbubbles on the stone surface cause twinkling (Lu *et al.* 2013). Yet the hypothesis is not fully accepted, partly because bubbles were not observed. Here, we extend the overpressure results to include under-pressure and use high-speed photography to visualize the bubbles. A programmable ultrasound system with Philips/ATL P4-2 transducer was used. *Ex vivo* COM stones were placed in a hydraulic pressure chamber and imaged acoustically through an acrylic window. The overpressure threshold to diminish twinkling was found to vary significantly, with twinkling eliminated at pressures of 3 ATA (atmospheres absolute) up to >8 ATA, even within the same stone. When the stones were exposed to 0.2 ATA (under-pressure), twinkling increased. High-speed photography during Doppler ultrasound revealed only one instance of an oscillating bubble. However, when stones were exposed repeatedly to a pre-focal, off-axis lithotripter pulse ($p_+ = 1.5$ MPa, $p_- = 2.5$ MPa), stones that twinkled had bubbles emerge from the same location with each pulse whereas stones that did not twinkle had a random bubble distribution. [Work supported by NSBRI through NASA NCC 9-58 and NIH DK043881, DK092197.]

3:15–3:30 Break

3:30

1pBA8. A new perspective for lung ultrasonography, preliminary results. Libertario Demi (Eindhoven Univ. of Technol., Den Dolech 2, Eindhoven 5612 AZ, Netherlands, l.demi@tue.nl), Wim van Hoeve (Tide Microfluidics, Enschede, Netherlands), Marcello Demi (Medical Imaging Processing, Fondazione Toscana Gabriele Monasterio, Pisa, Italy), Ruud J. van Sloun (Eindhoven Univ. of Technol., Eindhoven, Netherlands), Gino Soldati (Emergency Medicine Unit, Valle del Serchio General Hospital, Lucca, Italy), and Massimo Mischi (Eindhoven Univ. of Technol., Eindhoven, Netherlands)

Lung ultrasonography (LUS) is increasingly applied for the diagnosis of lung diseases. However, diagnoses are often based on imaging artifacts, e.g., B-lines, ultimately being qualitative and subjective. Aiming at gaining insight on the genesis of B-line artifacts, and on their link to the anatomical structures related to pathological conditions, dedicated lung-mimicking phantoms were manufactured and imaged with the ULA-OP research platform, proving access to raw radio-frequency (RF) data. To mimic a healthy and a diseased lung, two phantom types were made, being gelatin phantoms containing bubbly-layers composed of mono-disperse microbubble populations of different diameters: 140 and 80 μm, respectively. In fact, in various pathological conditions the size of the alveoli (air sacs composing the lung parenchyma), and hence the lung volume occupied by air, is alterated due to, e.g., fluid extravasation, alveolar collapse, and inflammation. Results show the appearance of B-lines only for phantoms designed to mimic pathological condition, confirming the link between artifact formation and reduction of air spaces dimensions. These preliminary results may be applicable to LUS, opening the way to the development of a quantitative ultrasound method dedicated to the lung.

3:45

1pBA9. Imaging of the femoral neck cortical bone based on iterative time domain topological energy. Chao Han, Didier Cassereau (Laboratoire d'Imagerie Biomédicale, Laboratoire d'Imagerie Biomédicale, Paris, France, chao.han@upmc.fr), Vincent Gibiat (Laboratoire PHASE, Toulouse, France), Jean-Gabriel Minonzio, Pascal Laugier, and Quentin Grimal (Laboratoire d'Imagerie Biomédicale, Paris, France)

Osteoporosis is a frequent bone disease that mainly affects women after menopause. It is characterized by a decrease in bone mass and a deterioration of the micro-architecture, which can lead to an increased risk of fracture. Ultrasound technologies provide an affordable mean to implement non invasive solutions to diagnostically assess the characteristics of the bone structure. In this work, we are interested in imaging the external and internal boundary of the cortical bone, and the evaluation of the thickness using the Time Domain Topological Energy (TDTE) method. These two properties are important to interpret the measurements of guided waves dispersion curves and predict the risk of osteoporosis and fracture. The iterative procedure of TDTE provides better performance in imaging the internal boundary of the cortical bone compared to the one-step TDTE. The barrier of implementing iterative TDTE is the requirements of huge storage during the numerical propagation step. A specific back-propagation strategy is implemented here to deal with this obstacle. The obtained results will be illustrated numerically and experimentally.

4:00

1pBA10. Development of complex tissue-mimicking phantoms for quantification of flow by the time-intensity method in contrast enhanced ultrasound imaging. Asawari Pawar, Gregory Clement, and Mark Howell (Biomedical Eng., Cleveland Clinic Foundation, Cleveland Clinic Foundation, Lerner Res. Inst., 2111 E 96TH St, Cleveland, OH 44106-2917, asawaripawar26@gmail.com)

Currently, there are few simple-to-construct *in vitro*, wall-less phantoms that have accurate acoustic properties while mimicking the complex normal and neoplastic geometries of the vascular network. The purpose of this study was to develop agar-based tissue-mimicking phantoms (TMP) to model such networks. Three types of vascular networks were considered: (1) single vessel, (2) multi-vessel with artery bifurcations, and (3) multi-vessel with artery bifurcations and structural abnormalities typical of diseased (tumor) vascular networks. Blood-flow related parameters were derived from the time-intensity curves obtained from the bolus injection of a lipid-based microbubble ultrasound contrast agent (UCA) under varying flow conditions relevant to our ongoing work in developing techniques to simultaneously quantify both the total volume and flow measurements within a tumor phantom. A Fukuda Denshi Ultrasound system was used with a linear probe (LG308-16A) positioned transversely and longitudinally to the direction of the flow. B-mode image acquisition was performed with 0.5 mL of UCA bolus injected into a 500 mL degassed water reservoir and then pumped through the vessels at rates ranging from 20 mL/min to 100 mL/min. Offline analysis of time-intensity curves in response to varied flow conditions indicated the TMP ability to yield easily reproducible simulations of vascular microcirculation.

4:15

1pBA11. Evaluating the robustness of an ultrasound based sensing strategy for intuitive control of upper extremity prosthetics. Nima Akhlaghi (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, nakhlagh@gmu.edu), Alex Baker (Bioengineering, George Mason Univ., Fairfax, VA), Huzefa J. Rangwala, Jana Kosecka (Comput. Sci., George Mason Univ., Fairfax, VA), and Siddhartha Sikdar (Bioengineering, George Mason Univ., Fairfax, VA)

Current commercially available prostheses based on myoelectric control have limited functionality, leading to many amputees abandoning use. Myoelectric control using surface electrodes has a number of limitations and lacks specificity for deep muscles, presenting a continued need for more robust strategies. We propose a new strategy for sensing muscle activity based on real-time ultrasound imaging. Results from our previous work demonstrate that complex motions could be classified with 92% accuracy in real-time. However, arm and hand repositioning during natural movements tend to alter the geometry of forearm musculature, possibly affecting performance. In this study, we evaluated the robustness of the image-based control strategy in the presence of varied forearm positions on able-bodied subjects. Ultrasound images of the forearm muscles were collected during two different scenarios using a Sonix RP with a 5–14 MHz linear probe. The subject was asked to perform four hand motions at eight different arm positions and three levels of wrist pronation. Images were analyzed to generate activity patterns for each motion and then classified. Results demonstrate that forearm positions do not significantly compromise reliability. We also show that performance could be further improved by including additional training activity patterns corresponding to motions performed in a few selected arm positions.

4:30

1pBA12. Kidney stone specific ultrasound imaging of human subjects. Bryan W. Cunitz, Barbrina L. Dunmire, Michael Bailey, Yasser Haider, Adam D. Maxwell, Julianna C. Simon (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, mike.bailey.apl@gmail.com), Jeff Thiel (Dept. of Radiology, Univ. of Washington School of Medicine, Seattle, WA), Oleg A. Sapozhnikov (Appl. Phys. Lab/Dept. of Acoust., Univ. of Washington/ Moscow State Univ., Seattle, WA), Jonathan D. Harper, and Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Sensitivity and specificity data show users have more difficulty in identifying stones accurately in B-mode ultrasound than x-ray CT. Our goal was to evaluate the signal to noise (SNR) of a new stone specific imaging algorithm, S-mode based on Color Doppler twinkling artifact, to B-mode. Forty sets of B- and S-mode imaging data were collected from 16 subjects using a Philips HDI C5-2 imaging probe and Verasonics ultrasound system. Two ways S-mode differs from Doppler is that it filters out blood flow signal and uses reverse color write priority to add color to echogenic regions only. For both B- and S-mode raw data, we calculate SNR of the magnitude (brightness) of the stone signal compared to the second highest magnitude in the image. The mean and standard deviation of the SNR was 1.6 ± 0.7 for B-mode and 37 ± 24 for S-mode, with 1 being the stone is equally bright as, and difficult to distinguish from, background. In this human study of S-mode, stones appeared over 30 times brighter than background and with over 20 times the contrast to background seen in B-mode. [Work supported by NIH NIDDK grants DK043881 and DK092197, and NSBRI through NASA NCC 9-58.]

4:45

1pBA13. Investigation of on skin surface response due to acoustic radiation from stenosed blood vessels. Huseyin Enes Salman and Yigit Yazicioglu (Mech. Eng. Dept., Middle East Tech. Univ., Orta Dogu Teknik Universitesi, Universiteler Mahallesi,, Dumlupinar Bulvari No:1, Cankaya, Ankara 06800, Turkey, salman@metu.edu.tr)

Arterial stenosis is a form of cardiovascular disease which leads to highest rate of fatalities worldwide. When stenosis is present in arteries feeding the heart, it leads to heart attack and often sudden death. If occlusion is present in arteries feeding brain, it causes stroke and rapid loss of brain functions. Arterial disease is usually not confined to critical organs such as the heart and the brain but observed throughout the peripheral cardiovascular system. Acoustic radiation due to stenosis propagates through soft tissues and reaches to skin surface. Therefore, these signals detected on skin surface may provide valuable information for diagnostic purposes. In this study, effects of acoustic radiation on skin surface due to a stenosis are investigated. Human upper arm is modeled using commercial finite element software ADINA. Realistic geometries and soft tissue mechanical properties are employed. Acoustic pressure distribution due to constriction is modeled using related studies in literature and applied on inner surface of blood vessel. Harmonic analysis is performed for upper arm and pressure distribution on skin surface is obtained. It is observed that increasing level of stenosis leads to an increase in pressure amplitudes on skin surface where the region which is closest to the constricted artery has the highest pressure amplitudes.

Session 1pNS

Noise and Animal Bioacoustics: Soundscape and Its Application

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Standardization in soundscape and its application. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de) and Bennett Brooks (BROOKS Acoust. CORP, Vernon, CT)

Since 2014, the first ISO Standard in Soundscape is on the market: IS0 12931-1, 2014 Acoustics — Soundscape — Part 1: Definition and conceptual framework. The Standard "explains factors relevant for measurement and reporting in soundscape studies, as well as for planning, design and management of soundscape." It is a first but big step to accept the idea that the perception of people has at least the same relevance as the physical measurements for the purposes of urban planning. Moreover, as sound is considered to be a resource and not a waste, the door is now open for designing our acoustic environment. The next step in this process is the development of ISO 12931-2 on methods and measurements, from which we will learn more about the character of holistic judgments. In addition, there is much new work on ANSI standards that consider life in park and wilderness areas. All of these engagements are directed to enhance the quality of life not only for humans but also for non-human beings. The paper will describe and discuss these different approaches.

1:25

1pNS2. Virtual acoustic environments for soundscape research and urban planning. Michael Vorlaender and Jonas Stienen (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

The utilization of Virtual Reality technology offers fascinating perspectives for the assessment of urban environments regarding the comfort, well-being, feeling of safety, and public health. Such technology, if available with a user-friendly software interfaces to existing and future planning tools, will expand the planning process into an integrative approach. Apart from sound, this may also involve vision and other aspects such as air pollution or climate change. The future design outline of the buildings, city quarters, and cities up to megacities can be assessed by taking the different perceptual modalities into account in an integrative procedure. Facilitating Virtual Reality for future environments in the assessment of new urban projects enables the population to get involved and renders the procedure more ecologically valid. Thus, by inclusion of the population, large counterforces, even demonstrations, or citizen's initiatives against urban or transport development (e.g., current discussions about infrastructure, airports, railway line planning, and wind farms) could be avoided. Recent developments in Virtual Reality technology for the built environment are highlighted by focusing on acoustics and noise control but also integrating vision, climate change, and other aspects of the multimodal indoor and outdoor assessment of comfort.

1:45

1pNS3. Auditory meaningscapes—An ecological perspective on soundscape perception. Frederik L. Nielbo (Dept. of Aesthetics and Commun., Ctr. for Semiotics, Aarhus Univ., Nordre Ringgade 1, Aarhus 8000, Denmark, norfln@dac.au.dk)

As a semiotic resource, the soundscape carries information about events relevant to the perceiving organism. Throughout evolution, the auditory system has been shaped to detect, localize, and identify such significant events in the environment in order to initiate appropriate behavior. The sounds are, in other words, environmental sign vehicles to be picked up and utilized by the perceiving organism, allowing it to navigate in the surroundings, avoid potential dangers, etc. Far from being a passive receiver, the auditory system is an active semiotic tool for collecting information relevant for the given situation. From the rudimentary hearing system of simpler organisms to the complex human auditory system, perception of auditory events is functional and instrumental; it guides the perceiver's behavior by carrying information about potential interactions with the environment. A growing number of studies suggest that auditory perception is intimately coupled with action and that listeners' perception of soundscapes is structured around semantic categories related to events and activities taking place in the heard environment. Thus, I argue, to get a richer understanding of soundscapes, it is necessary to pay attention to the semiotic dynamics of the interaction between the perceiver and the sonic environment.

2:05

1pNS4. Soundscape response in animals. Susan M. Wiseman (Waco, TX, sw1210txstate@gmail.com)

The World Health Organization (WHO) has long warned against chronic or extreme noise exposure as it has been shown to impact humans, as have the U.S. Surgeon General and a variety of academic and health research studies. Little is known about safe sound levels for animals, let alone safe frequency exposure for specific species. Behavioral and physiological response has been noted in human and non-human animals, most obviously fright and flight in the face of major stimuli such as thunder or gunshots at one end of the spectrum, down to minute stimuli such as the rustle of a leaf indicating the proximity of prey or a predator. Some animals are attracted or deterred by certain sounds, some mimic, in agriculture some become more or less productive according to their soundscape. To improve animal welfare, the Association of Zoos and Aquariums requires the enrichment of captive environments with the goal of increasing an animal's behavioral choices and drawing out species-appropriate behavior. It has been noted that certain southern white rhinoceros *(Ceratotherium simum simum)* altered their behavior in response to a variety of sound stimuli, including music, illustrating how the soundscape can be manipulated as a tool for animal enrichment.

2:25

1pNS5. Patient centered method and soundscape—A bridge between clinicians and acousticians. Robert Y. McMurtry (Surgery, Schulich School of Medicine, Western Univ., 403 Main St., Picton, ON K0K2T0, Canada, rymcmurtry1@gmail.com)

The Patient Centered Method (PCM) and Soundscape have much in common including their emergence about 60 years ago based on the work of Balint and Kryter, respectively. Both place the patient or person at the center of management of clinical illness or noise annoyance. PCM requires that the patient perceive that they have experienced meaningful care, communication, and common ground in clinical encounters. The evaluation focuses on the patient's life context and their perception of disease or the "illness experience." When PCM is accomplished, the result is higher satisfaction, better outcomes of chronic diseases, fewer tests, and referrals and attendant lower costs (Stewart *et al*., 2000). Soundscape, a term coined by Shafer in 1977 also places the person in center, in the context of their acoustic environment, emphasizes their perception of noise as the "New Experts" (Bray 2012). According to Bray exposed people are "objective measuring instruments whose reports and experiences must be taken seriously and quantified by technical measurements." This paper will explore the congruence of PCM and Soundscape and the necessity of this approach in evaluating the experience of those exposed to wind turbine acoustical energy.

2:45

1pNS6. Acoustical impact of wind turbines on soundscape. Klaus Genuit, Andre Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de), and Brigitte Schulte-Fortkamp (Tech. Univ. Berlin, Berlin, Germany)

It is well known that wind turbines have a negative impact on the landscape. But what is their impact on soundscape? Is this independent from the visual perception? Increasing research efforts are made to explain annoyance and complaints caused by wind turbines in detail, but still several questions are unanswered. The acoustical contribution of wind turbines depends on the technical design of the generator and of the blades. All of them produce low frequencies; some of them produce noise in the middle and higher frequencies with tones and modulations. It is clear that the A-weighted sound pressure level is not the appropriate indicator to predict resulting annoyance. The question is what must be considered to understand the perceived sound quality within the context of soundscape? Are psycho-acoustical parameters able to describe the sound character of wind turbines in a better way? To get an improved understanding of the complex interactions of the sound produced by wind turbines and the existing sounds of a given soundscape basic studies were performed and the results will be presented.

3:05–3:20 Break

3:20

1pNS7. Soundscape of a wind farm—The Cape Bridgewater experience. Steven E. Cooper (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Namibia, drnoise@acoustics.com.au)

The general concept for describing the noise environment in proximity to a wind farm is expressed in terms of the A-weighted level that will vary dependent upon the wind strength. The compliance methodology in general use for wind farms relies upon a measurement that includes wind and an average line of fit through such data. Measurements at residential receivers when conducted using full spectrum recording/analysis revealed unique characteristics extending into the infrasound region that are normally inaudible that would appear to be present when disturbance is noted. There are no traditional dose-response investigations for the full spectrum of wind farm noise on which to describe the soundscape. A new approach to assessing wind farm noise emissions was used for the Cape Bridgewater wind farm study identifying different concepts for describing the soundscape of the wind farm

3:40

1pNS8. Uses of soundwalks in computer modeling of soundscapes during the design process. Gary W. Siebein, Marylin Roa, Gary W. Siebein, Hyun G. Paek, and Paul C. Jones (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

This paper presents a case study of the use of soundwalks to gather data for computer simulations of complex soundscapes while they are being designed. A soundwalk is a process where people enter an existing soundscape for listening, observation, measurement, recording or evaluation. This project is for a new mixed use urban center. Studies were undertaken to investigate the soundscape of this new community both within the community itself as well as outside the community while it was being designed. Soundwalks were taken through existing communities with similar buildings, urban spaces and infrastructure to those being proposed in the design. Acoustical measurements made during the soundwalks of specific acoustic events that occurred in the communities were used as input data for

1p MON. PM

sound sources in a large, three dimensional computer model of the site and its environs. The model was used to estimate future sound levels from the mix of activities in the new community. Aural recordings of the specific acoustic events were used in aural simulations of the spaces where stakeholders could evaluate the acoustical conditions in the community and provide insights into the design of the soundscape.

Contributed Papers

4:00

1pNS9. Water soundscape and listening impression. Yosua W. Tedja and Lucky Tsaih (Architecture, National Taiwan Univ. of Sci. and Technol., RB-807, No.43, Sec. 4, Keelung Rd., Da'an Dist., Taipei 10607, Taiwan, yosua_wiranata@yahoo.co.id)

This paper presents the preference of water soundscape through listening evaluation. As meditation, listening to water sound is a tool that often use to provide positive emotion and psychological restoration. Twenty water soundscape samples were chosen based on natural and man-made sounds, as well as the sound in relation to the architecture and materiality. A semantic differential questionnaire was created with nine pairs of contractual sound qualities. Sixty-six architectural students with normal hearing condition were participated in this listening evaluation. The result has shown that 92% of the participants have positive impression to a soft quiet mid frequency sound (gentle stream) due to the listening impression of relax (88%) and comfort (89%) qualities. On the contrary, 77% of the participants have a negative impression for a loud intense broadband sound (rain on glass roof) due to the listening impression of noisy (82%) and agitating (80%) qualities. Sixty-two percent of the participants prefer rain on water sound due to the listening impression of comfort (60%) more than rain on different materials such as woods, metal, glass, tent, and pavement. Shishiodoshi and Suikinkutsu were also identified with "quiet" sound quality but overall likely impression is not as high as water fountain.

4:15

1pNS10. Comparison of urban and rural soundscapes associated with *Dumetella carolinensis* and *Cardinalis cardinals*. David P. Knobles (Knobles Sci. Applications LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, Delaware, DE), and Preston S. Wilson (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Both the acoustic transmission properties of the environment and the ambient noise field are factors that affect avian soundscapes. Long-term acoustic measurements made with Song Meter SM2+ recorders in both Texas and Delaware are analyzed for the information they contain on soundscapes as they pertain to songbirds, such as *Dumetella carolinensis*, commonly found in Delaware, and *Cardinalis cardinals*, common to both Delaware and Texas. The measurements in both states are made in both urban and rural environments and in coincidence with physical measurements of temperature, humidity, wind speed, and direction. Additional acoustical measurements are made with other sound recorders to gain more specific information about, for example, the dawn chorus. The efficacy of using the characteristics of the chorus as an indicator of the health of the avian soundscape is discussed. Band limited cross-correlation methods are used to test various hypotheses on bird communication and behavior.

4:30

1pNS11. What do hedonic studies of the costs of road traffic noise nuisance tell us? Abigail Bristow (School of Civil and Bldg. Eng., Loughborough University, Loughborough LE11 3TU, United Kingdom, a.l.bristow@lboro.ac.uk) and Sotirios Thanos (Univ. College London, London, United Kingdom)

The value or cost of noise nuisance is important as it enables judgments to be made on the costs and benefits of interventions. This paper presents a review and meta-analysis of hedonic pricing (HP) studies of road traffic noise nuisance. Noise nuisance has commonly been valued using HP, a revealed preference approach based on the housing market, where house price is a function of a myriad of characteristics of the house and the surrounding area including noise. The value of noise obtained is expressed as the percentage change in house prices that results from a 1 decibel (dB) change in noise levels (Noise Depreciation Index, NDI). The approach is broadly accepted and underpins most values used in Government transport appraisals. However, the range of values is large, from 0.08 to 2.21 NDI according to the last review of the literature in 2001. This paper examines available studies to shed light on the variation in noise values, as well as new methodological developments, such as the widespread use of spatial econometrics and addressing non-linearities of noise values.

4:45

1pNS12. Implementation of an augmented reality interface to reproduce and compare soundscapes. Philip W. Robinson (Specialist Modelling Group, Foster + Partners, 22 Hester Rd., London SW11 4AN, United Kingdom, philrob22@gmail.com)

Soundscape design is a growing concern in architectural practice; yet, conveying soundscape properties is a challenge. Quantitative measures like decibel levels offer little utility; birdsong and squealing brakes may be at the same frequency and level, yet one is obviously preferable. Furthermore, visual qualities of the scene have a substantial effect on perception of the acoustic environment. To improve communication between acoustic designers, architects, and clients, an augmented reality interface has been designed to allow comparisons of soundscapes and their relationships to the built environment. The interface consists of a physical scale model, printed data, and a tablet application. The augmented reality application uses a machine vision algorithm to recognize the model and allow on-screen interaction. On the tablet, measurement points are displayed on a live image of the model at their respective locations, and the user may select a point by touch, which will begin sound playback and display an immersive image of the scene. The participant can then look around a spherical image of the scene using the tablet as a movable window, and thereby listen to the environment with appropriate accompanying visual cues. The interface has been a useful tool to communicate urban sound issues.

5:00

1pNS13. Residents' sound preference of rural soundscape in China. Xinxin Ren (School of Architecture, Harbin Inst. of Technol., 66 Xida Zhi St., Nangang District, Harbin 150006, China, xinxin088521@126.com), Jian Kang (School of Architecture, Harbin Inst. of Technol., Sheffield, United Kingdom), and Hong Jin (School of Architecture, Harbin Inst. of Technol., Harbin, China)

The importance of rural soundscape has been recently recognized for the recreational and amenity value, where man-made noise is at a low level and the dominance of natural sounds promotes sound preference, but there is a lack of studies on the sound preference of residents in the villages of China, which are undergoing a fast development in urbanization. In this study, a questionnaire survey was undertaken at 36 villages of northeast regions, representing all major soundscape types. The results show that among the individual factors, age, education level, and landscape environment are more significant than the factor of gender, in terms of influence on sound preference. With increasing age, the sound preference of traffic and mechanical sound has a tendency of increase and then decrease, whereas the tendency for human voice, natural, and melody sound is opposite. With a higher education level, the sound preference of traffic, mechanical sound, and human voice decreases, and on the contrary, sound preference for natural and melody sound increases. However, the evaluation of natural and melody sound is lower in better landscape environment, which is perhaps due to the residents' higher requirements in soundscapes corresponding to the landscapes.

5:15–5:35 Panel Discussion

Session 1pPA

Physical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Phononic Metamaterials II

Joel Mobley, Chair
Physics and Astronomy, University of Mississippi, PO Box 1848, 1034 NCPA, University, MS 38677

Contributed Papers

1:00

1pPA1. Expanding the low frequency transparency band of shell structures for sonic crystals. Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, alexey.titovich@navy.mil), Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, MD), and Stephen D. O'Regan (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD)

Elastic cylindrical shells are often used as the elements of sonic crystals for wave steering applications. The thickness-to-radius ratio of the shell determines its quasi-static acoustic properties: effective bulk modulus and density. Matching the bulk modulus to that of the surrounding fluid removes the monopole response, while matching the density removes the dipole response. Together, they yield a low frequency band in which the shell is acoustically transparent. In this study, we look to broaden the transparency band by removing the quadrupole response as well. Among other methods, we will investigate the potential for two nested, counter-vibrating, shells to control quadrupole radiation.

1:15

1pPA2. Propagation of pulsed ultrasonic fields in a band gap of a two dimensional phononic crystal. Ukesh Koju and Joel Mobley (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677, ukoju@go.olemiss.edu)

A band gap in the transmission spectrum of a finite two dimensional phononic crystal is examined in the time domain using pulsed ultrasonic fields. The phononic crystal consists of a hexagonal array of copper cylinders ($r = 1.19$ mm) in an aqueous matrix with a lattice constant of 2.9 mm. Measurements of the transmission properties of the sample are performed using ultrasonic wave groups of various center frequencies and bandwidths. Among the band gaps in the low-MHz range, we concentrate on the gap from 1.48 MHz—1.70 MHz. The phase velocity, group velocity, and attenuation coefficient spectra are determined and compared with expectations.

1:30

1pPA3. Acoustic scattering cancelation in an aqueous environment using phononic crystals. Matthew D. Guild (NRC Res. Associateship Program, Naval Res. Lab, Washington, DC), Theodore Martin (Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, theodore.martin@nrl.navy.mil), Charles Rohde, David Calvo, and Gregory Orris (NRC Res. Associateship Program, Naval Res. Lab, Washington, DC)

Acoustic scattering cancelation is an approach that enables the elimination of the scattered field from an object within the surrounding medium. While this cancelation effect can be achieved through the use of a single, isotropic fluid, such a simple design limits its application to objects that are small compared to a wavelength. More complicated multilayered fluidic coatings are necessary for the cancellation of higher order scattering modes. Such modes arise from objects with characteristic lengths comparable to the incident wavelength in the surrounding medium. To realize such a coating, phononic crystals offer a means for precisely designing the necessary structures. In this work, analytical and numerical results will be presented for the acoustic scattering cancellation of cylindrical objects in an aqueous environment using multilayered effective fluid coatings constructed of phononic crystals. [Work supported by the Office of Naval Research and the National Research Council.]

1:45

1pPA4. Negative refraction of acoustic waves in phononic crystals using recursive algorithms for block toeplitz matrices. Feruza A. Amirkulova (Phys. and Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, feamirkulova@vassar.edu) and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Recent improvements in design and manufacturing have significantly improved our ability to manage phononic crystals (PC). Researchers have been studied the negative refraction in PC experimentally and theoretically using multiple scattering (MS) theory. This problem requires solving a large complex valued linear system that has a special multilevel block Toeplitz (BT) structure. We study negative refraction of acoustic waves in 2D PC by means of MS theory, by taking advantage of the PC structure and using specific recursive algorithms for BT matrices. We present new efficient and accurate algorithms for solving acoustic MS problem by the cluster of closely spaced cylinders to design acoustic negative refraction imaging in a PC. The unit cell of the PC consists, in general, of a solid cylinder in an acoustic medium. The dispersion curves of PC have a negative refraction dispersion branch producing the focusing effect. Particular attention is given to the dynamic behavior in the vicinity of dispersion branches with negative slope, and to the band gaps. We explore the effect of focal point on structural parameters, and employ a parallelization technique that allows efficient application of the proposed recursive algorithms for solving BT systems on high performance computer clusters. Numerical comparisons of CPU time and total elapsed time taken to solve the linear system using the direct LAPACK and TOEPLITZ libraries on Intel FORTRAN show the advantage of high performance recursive algorithms over the Gaussian elimination.

2:00

1pPA5. Acoustic phase hologram with labyrinthine metamaterials. Yangbo Xie (Elec. and Comput. Eng., Duke Univ., 3417 CIEMAS, Durham, NC 27705, yx35@duke.edu), Chen Shen, Yun Jing (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Acoustic metamaterials offer large degree of design freedom and precise control over amplitude and phase at subwavelength scales. In the past, we have demonstrated a family of labyrinthine metamaterial unit cells as premium building blocks for phase modulation devices, as well as several 1D wavefront shaping devices based on these unit cells. Here we extend the complex modulations to 2D by demonstrating a computer generated metamaterial-based phase hologram. Through spatially modulating the phase of the wavefront, the hologram projects the incident wave to a designed three-dimensional amplitude pattern. The hologram is designed with a two-step

process: first, an iterative holographic reconstruction algorithm is used to obtain the optimal phase pattern of the hologram; second, unit cells with desired phase modulations are designed and fabricated. Ray tracing and full-wave simulations have been performed to verify the design. The measurements of reconstruction in an anechoic chamber will be taken for a hologram designed to operate around 4 kHz. The metamaterial-based hologram creates a three-dimensional acoustic illusion with only passive structures. The designing process can also be extended to devices such as multi-focal lenses and wave-based analog processing/computing interfaces.

2:15

1pPA6. Metascreen-based acoustic passive phased array with subwavelength resolution. Likun Zhang (The Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu), Yong Li (CNRS-Université de Lorraine, Vandœuvre-lés-Nancy, France), Xue Jiang, Bin Liang, and Jian-chun Cheng (Nanjing Univ., Nanjing, China)

A phased source array is an array consisting of elementary sources with proper relative phases to steer a wavefront, so as to form desired wave fields of specific property and applications. However, the phased array requires a large number of sources in forming complex wavefront or non-paraxial wave beams, leading to high cost and complexity in the electronics required to operate individual sources of the active array. A passive metascreen is presented here to transmit sound energy from a single source and steer the transmitted wavefront to form desired fields. The metascreen plays a role like a phased array but with a passive way that avoids the complexity of an active array. The screen has a half-wavelength thickness and composes of a series of elements with a dimension of one-tenth of the sound wavelength along the screen. The elements have a hybrid structure designed for high transmission and full range of phase shift. The performance of the screen is numerically simulated and experimentally demonstrated to generate a self-bending beam in non-paraxial region. The screen with its simple configuration and extreme acoustic performance could have applications for sound field shaping in numerous areas where the conventional array would have complexity and limited capability.

2:30

1pPA7. Subwavelength acoustic metamaterial with tunable acoustic absorption. Nicolas Viard, Cali Gallardo, Jun Xu, and Nicholas Fang (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, nviard@mit.edu)

We present numerical simulations and experimental measurements for ultrasonic transmission through a subwavelength metamaterial consisting of an array of hollow cylinders embedded in a soft elastic matrix. The mechanical properties of the matrix, the lattice constant, and the size of the cylinders are optimized in order to maximize sound absorption in the metamaterial while the cylinders are filled with air. The acoustic transmission is restored when the cylinders are filled with water. Our design expands the concept of metamaterial as it demonstrates the ability to tune the acoustic properties of a subwavelength material.

2:45

1pPA8. Rotating the directivity of sound radiation by acoustic metastructures. Likun Zhang (Dept. of Phys. and Ctr. for Nonlinear Dynam., The Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu), Xue Jiang, Bin Liang, Xin-ye Zou, and Jian-chun Cheng (Key Lab. of Modern Acoust., MOE, Inst. of Acoust., and Collaborative Innovation Ctr. of Adv. Microstructures, Nanjing Univ., Nanjing, China)

Radiation directivity is an important measure of sound fields radiated from acoustic sources. Manipulation of the directivity plays a significant role in many situations ranging from audio and auditorium acoustics to medical ultrasound applications. An acoustic meta-structure is presented here to rotate the directivity of sound radiation from sources surrounded by the meta-structure with an anisotropic acoustic property. The meta-structure consists of an array of platelets, regularly arranged in several concentric rings. The platelets' orientation is determined from transformation acoustics model. Numerical simulations of sound fields with the structure reveal that the angle of sound directivity rotated by the structure is independent of sound frequency and source location. The rotation is verified through experimental measurements with the meta-structure fabricated by thermoplastics materials via 3D printing. The rotation angle varies over a broad range through tuning geometry parameters of the meta-structure. The meta-structure with its extreme performance can hence provide various applications in the effective control of radiation directivity in acoustic engineering.

Session 1pSP

Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography: Direction of Arrival (DOA) Estimation, Source Localization, Classification, and Tracking Using Small Aperture Arrays II

Geoffrey H. Goldman, Cochair
U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

Invited Paper

1:30

1pSP1. Tomographic sonar image formation of difficult objects and of mines on the seabed. Brian G. Ferguson (DSTO, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au) and Ron J. Wyber (Midspar Systems, Oyster Bay, NSW, Australia)

Insonifying a sea mine over a complete (360°) set of look directions, while recording the acoustic returns (or echoes) from the object as a function of aspect angle, enables the two dimensional projection data (or measurement) space to be fully populated. An acoustic image is readily reconstructed by applying Fourier transform methods to the projection data. Tomographic sonar images of various (inert) sea mines are presented, where a fixed monostatic sonar insonifies the object as it rotates about its vertical axis through one complete revolution. A difficulty arises if the specular component returned from one part of the object swamps the returns from the rest of the object. The problem is solved by taking spatial derivatives of the Fourier reconstructed image, which then results in a representation that enables the object to be recognized. Another complication presents itself when structural waves, which are excited by the insonification process, contribute to the observed impulse response. Numerous examples of the effect of these structural waves on the formation of the image are presented for various objects and sonar technologies, including parametric sonar. For operational tomography, the object is fixed (a mine resting on the sea floor), which requires the sonar to circumnavigate the mine (at a safe standoff distance) while simultaneously insonifying it and compiling the multiaspect projection data space. Sample tomographic sonar images of a practice sea mine, a Mk 84 bomb, and the sea floor itself, are presented.

Contributed Papers

1:50

1pSP2. Fin and humpback whale vocalization classification and localization, northern Georges Bank. Wei Huang (Elec. and Comput. Eng., Northeastern Univ., 500 Broadway, Apt. 3157, Malden, MA 02148, huang.wei@husky.neu.edu), Delin Wang, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Several tens of thousand vocalizations from multiple fin whale individuals were passively recorded by a high-resolution coherent hydrophone array system in the Gulf of Maine in Fall 2006. The recorded fin vocalizations have short durations roughly 0.4 s and frequencies ranging from 15 to 40 Hz. The fin vocalizations were detected via spectrogram intensity thresholding. The horizontal azimuth or bearing of each detected fin whale vocalization was determined by broadband beamforming. Each vocalization was then characterized using numerous features, such as center frequency, upper and lower frequency limits, and duration, obtained from pitch tracking. These vocalizations were then classified using Bayesian-based Gaussian Mixture model feature clustering into several distinct vocal types. The vocalization clustering result was then combined with the bearing-time trajectory and localized by the moving array triangulation and the array invariant techniques. The vocalization types are found be dependent on the geographic region, suggesting a potential application for monitoring different fin whale groups from their vocalization features. A similar approach can be used to monitor humpback whale groups.

2:05

1pSP3. Micro-aperture bio-inspired broadband sonar model and system for underwater imaging applications. Jason E. Gaudette (Sonar and Sensor Systems, NUWC Div. Newport, 1176 Howell St., B1320/135, Newport, RI 02841, jason.e.gaudette@navy.mil), Philip Caspers (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

Conventional angular sonar imaging is based principally on the correlation of signals received across individual elements in an array. Thus, array signal processing simply matches a particular set of time delays (or equivalently, phase shifts) to the corresponding arrival angles. The angular resolution of such conventional systems is fundamentally limited by the aperture-to-wavelength ratio. The biosonar of echolocating bats and dolphins provide inspiration that we can significantly overcome these aperture-to-wavelength limits by two orders of magnitude. The key to success lies in the use of multiple octaves of bandwidth. Previous work in bat echolocation has shown how multiple overlapping echoes in range can be deconvolved through spectral pattern matching of broadband interference notches. Recent modeling and simulation results show how these bio-inspired broadband interferometric techniques can be extended to the angular imaging problem despite having only two elements spaced at 1 to 4 λ. Acoustic tank testing results from an underwater prototype array will also be presented. [Work supported by ONR 341 and internal investments by NUWC Division Newport.]

Invited Papers

2:20

1pSP4. Hostile fire detection using a bio-inspired mobile acoustic sensor network. George Cakiades (US Army ARDEC, 407 Buffington Rd., Picatinny Arsenal, NJ 07806, george.cakiades.civ@mail.mil), Socrates Deligeorges (BioMimetic Systems, Cambridge, MA), Jemin George (Army Res. Lab, Adelphi, MD), Felipe Núñez (Univ. of California, Santa Barbara, Santa Barbara, CA), Yongqiang Wang (Clemson Univ., Clemson, SC), and Francis J. Doyle (Harvard Univ., Cambridge, MA)

Hostile Fire Detection (HFD) sensors play an increasing role in combating asymmetric threats in both military and civilian operations. Bio-inspired advances in acoustic sensor technology have enabled small aperture arrays to localize and identify target sounds on baselines as small as 7.5 cm, making them practical for body worn and mobile applications such as UAVs. The unique approach to acoustic processing reduces acoustic information through a neural transform to key features that allow segregation of multiple targets using spectro-temporal cues creating auditory objects. The sparse representation of targets as auditory objects enables fast computationally efficient localization, identification, and tracking of several acoustic targets nearly simultaneously. The sparse representation is also ideal for information fusion among sensors over limited bandwidth networks for enhanced performance in challenging environments. Through a collaborative effort between Army research groups, the University of California Santa Barbara, and BioMimetic Systems, a prototype acoustic sensor network using biologically inspired sensors and network synchronization for a mobile HFD application has been developed. The network employs body worn PinPoint^{TM} HFD sensors interfaced with smartphones (Android) running net-centric fusion algorithms. The network will be discussed in terms of the biologically inspired components, information fusion, as well as results from preliminary field tests.

2:40

1pSP5. Green's function retrieval for atmospheric acoustic propagation. Sandra L. Collier, Jericho E. Cain, John M. Noble, W. C. K. Alberts, David A. Ligon, and Leng K. Sim (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil)

There is an extensive classical utilization of the Green's function for wave propagation in many different media. By extracting the Green's function, or medium impulse response, one may obtain information about the medium channel. This information could be used to overcome the medium effects, as is done in time-reversed acoustic localization or acoustic communications; alternatively, it may be used to deduce information about the medium. The use of time-reversal methods has been established for interferometry, phase conjugation, and time-reversal mirrors and cavities. The objective of this research is to determine the feasibility of using a time-reversal method to extract a Green's function for outdoor acoustic propagation utilizing sources in the audible frequency ranges. Initial findings are presented here.

3:00–3:15 Break

Contributed Papers

3:15

1pSP6. Matched-field source localization with multiple small-aperture arrays. Dag Tollefsen (Norwegian Defence Res. Establishment (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no) and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper considers combining information from multiple small-aperture arrays in matched-field processing (MFP) for source localization. Assuming individual arrays are comprised of calibrated sensors which are synchronized in time, conventional MFP can be applied for each array and the resulting Bartlett processors summed over arrays. However, if the relative calibration and/or time synchronization is known between some or all arrays, more informative multiple-array processors can be derived by maximum-likelihood methods. For example, if the relative calibration between arrays is known, the observed amplitude variations between arrays provide additional information for source localization; if synchronization is known, phase variations provide localization information. Various multiple-array processors are derived and evaluated in terms of the probability of correct localization from Monte-Carlo analyses for a range of signal-to-noise ratios and number of frequencies for simulated shallow-water scenarios with vertical and horizontal arrays. Effects of environmental mismatch in seabed geoacoustic parameters and water depth are also considered. The analysis indicates that, dependent on array configurations, substantial improvements in source localization performance can be achieved when including relative amplitude and/or phase information in the multiple-array processor. The improvement is reduced by environmental mismatch; this degradation can be partially mitigated by including additional frequencies in the processing.

3:30

1pSP7. Acoustic tracking of small aircraft within an airport. Alexander Sedunov, Alexander Sutin, Nikolay Sedunov, Hady Salloum, Alexander Yakubovskiy (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu), and David Masters (Sci. and Technol. Directorate, Dept. of Homeland Security, Washington, DC)

The Acoustic Aircraft Detection (AAD) system developed by Stevens Institute of Technology for the detection of small aircraft in remote border areas was tested for tracking small aircraft in the airport. The goal of the test was to demonstrate a proof-of-concept capability for the acoustic sensor when applied to tracking aircraft during take off, landing, or taxing in a small airport. Two AAD nodes with five microphones were used to find the direction of arrival of sound produced by an aircraft using time-difference of arrival to pairs of sensors and to localize those by triangulation in real time. Additionally, a portable acoustic recorder system (PARS) with three microphones arranged in a ground plane was deployed to provide more data for post-processing analysis. The nodes separation varied between 150 and 250 m in different experiments. During the two-day experiment, aircraft activity of regular airport traffic was observed. At least 14 different aircraft were recorded during various movement on airport runways as well as takeoff and landing. The tested AAD was developed for another mission and was far from an optimal acoustic system for small airport applications. Suggestions for this system will be presented. [This work was sponsored by DHS S&T.]

3:45

1pSP8. Nearfield acoustic holography in noisy environment based on the measurement using an acoustic mask with microphone array. Shang Xiang and Weikang Jiang (Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, wkjiang@sjtu.edu.cn)

Nearfield acoustic holography (NAH) has been widely used for identifying acoustic noise sources. However, conventional NAH is theoretically available only in the free field; therefore, the NAH may meet difficulties in many application *in-situ* environments. An NAH based on the measurement using an acoustic mask with microphone array is proposed in this presentation to reconstruct sound pressure and velocity of target sources in the environment with reflections and disturbing acoustic noise sources. The microphone array is flush-mounted at the bottom plane of an open rectangular mask. Four side surfaces are designed to acquire sound pressure in Neumann's boundary condition. In an actual measurement, the mask faces to near field of a target source. Although the sound wave from disturbing sources may propagate into the mask by passing through the gap between the mask and source surface, the disturbing waves can be decomposed by the proposed NAH using inverse patch transfer functions. Numerical simulations and experiments indicate that this novel NAH using mask with microphone array is valid for reconstructing sound source in noisy environment. The influence of different distances between the source surface and the mask is also investigated.

4:00

1pSP9. Detection results for a class 1 unmanned aerial vehicle measured with a small microphone array. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

The Department of Defense is developing low size, weight, power, and cost (SWaP-C) acoustic systems to detect and track small unmanned aerial vehicles (UAV). To support these goals, an analysis of several detection algorithms was performed using acoustic data generated with a class I unmanned aerial vehicle and measured with a small tetrahedral microphone array. The detection algorithms were based upon the peak output power and the coherence factor of several beamforming algorithms. Receiver operation characteristics (ROC) curves were generated using both an energy-based and a Neyman-Pearson-based detection algorithm.

4:15

1pSP10. Atmospheric compensation of acoustic signatures of aircraft. Minas Benyamin and Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

The U.S. Army Research Laboratory is investigating using low size, weight, power, and cost (SWaP-C) acoustic sensors to classify unmanned aircraft systems (UAS). One issue complicating UAS classification using acoustics is that the atmosphere attenuates the signature of the targets in a complex manner. To address this issue, we developed and tested a technique to mitigate the effect of atmospheric attenuation. We used Bass's model with inputs from temperature, relative humidity and range to estimate atmospheric attenuation, then we used techniques based upon a Wiener filter implemented in the frequency domain to adjust the amplitude of the measured signal. The Wiener filter reduced the amplification of the noise while improving the reproducibility of the signature at different ranges, particularly at higher frequencies. The results suggest that preprocessing the data using this technique should improve the performance of acoustic classifier algorithms.

MONDAY EVENING, 2 NOVEMBER 2015 GRAND BALLROOM FOYER, 5:30 P.M. TO 7:00 P.M.

Exhibit and Exhibit Opening Reception

The instrument and equipment exhibit is located near the registration area in the Grand Ballroom Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems and other exhibits on acoustics.

The Exhibit will open on Monday with an evening reception with lite snacks and a complimentary drink.

Exhibit hours are Monday, 2 November, 5:30 p.m. to 7:00 p.m., Tuesday, 3 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 November, 9:00 a.m. to 12:00 noon.

Coffee breaks on Tuesday and Wednesday mornings (9:45 a.m. to 10:30 a.m.) will be held in the exhibit area as well as an afternoon break on Tuesday (2:45 p.m. to 3:30 p.m.).

The following companies have registered to participate in the exhibit at the time of this publication:

Brüel & Kjær Sound & Vibration Measurement—www.bksv.com

Freudenberg Performance Materials—www.Freudenberg-pm.com

G.R.A.S Sound & Vibration—www.gras.us

PCB Piezotronics—www.pcb.com/

Sensidyne—www.sensidyne.com

Springer—www.Springer.com

Teledyne Reson—www.teledyne-reson.com

Payment of an additional fee is required to attend.

MONDAY AFTERNOON, 2 NOVEMBER 2015 GRAND BALLROOM 6, 7:00 P.M. TO 9:00 P.M.

1eID

Interdisciplinary: Tutorial Lecture on Sonic Booms: A "Super" Sonic Saga

James P. Cottingham, Chair
Physics, Coe College, 1220 First Avenue, Cedar Rapids, IA 52402

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Interdisciplinary tutorial on sonic booms: A "super" sonic saga. Victor W. Sparrow (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

This tutorial will provide an introduction to sonic booms, including background, current status of research, and future prospects. Supersonic aircraft generate sonic booms, so the physics of shock propagation through the atmosphere is involved. But those shocks are ultimately heard on the ground indoors and outdoors, both by people and wildlife. Hence, physical acoustics, psychological and physiological acoustics, animal bioacoustics, structural acoustics, vibration, and noise are all involved. Aircraft manufacturers have plans to build new civilian supersonic passenger aircraft, and these new aircraft intend to have sonic booms that are much quieter than those produced by either military aircraft or the now-retired Concorde supersonic airliner. Thus, an additional objective of this tutorial is to provide a quick-start technical basis for ASA members to understand the basics of sonic booms, enabling them to communicate effectively about sonic booms. If the aircraft manufacturers meet their stated goals, millions of people around the world will potentially begin hearing quiet sonic booms in the future. The material is highly interdisciplinary and should be useful both to the curious and to specialists across almost all of the Acoustical Society of America's technical areas. [The opinions, findings, conclusions, and recommendations stated here are those of the author and do not necessarily reflect the views of sponsors of the ASCENT Center of Excellence including the Federal Aviation Administration.]

In conjunction with the Tutorial Lecture on sonic booms on Monday evening, Gulfstream Aerospace Corporation will be bringing their sonic boom simulator to the Jacksonville, FL meeting. The Supersonic Acoustic Signature Simulator II is a specially equipped mobile audio booth designed to accurately reproduce the noise an observer on the ground would hear if a supersonic aircraft flew by. More specifically, the visitor at the ASA meeting will experience a back-to-back comparison of two radically different synthesized sonic booms. The first sonic boom signature will represent the "traditional" N-wave signature produced by the Concorde. The second sonic boom will represent a shaped signature that is representative of a low-boom aircraft. The intent of this demonstration is to provide the visitor with an opportunity to experience a fully immersive simulation that contains a sophisticated auralization of stimuli that contribute to subjective response caused by a sonic boom.

The simulator will only be available on Monday, November 2 (12:00 noon to 7:00 p.m.), and Tuesday, November 3 (9:00 a.m. to 6:00 p.m.), and will be located at Jacksonville Landing, just steps away from the Hyatt Regency. Walking directions will be provided on site via convenient signs. Please come and take advantage of this special opportunity to hear the Supersonic Acoustic Signature Simulator II.

Session 2aAA

Architectural Acoustics and Noise: Acoustics of Multifamily Dwellings

Eric L. Reuter, Cochair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Effects of speaker location and orientation on room sound fields and room to room noise reduction. Noral D. Stewart and W. C. Eaton (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, noral14@sacnc.com)

ASTM E336 requires that loudspeakers be placed at least 5 m from a partition under test, and if that is not possible, then in the corners opposite the partition. The standard further advises that if directional loudspeakers are used in the corners, they should be faced into the corners. In practice, speakers are usually placed in the corners of residential and smaller office spaces, but not in the corners of large spaces. Recent experience has uncovered two effects that need further study. First, when loudspeakers are faced into the corners of a large room, the sound spectrum is strongly colored with a large dip typically in the 100 to 500 Hz range. This effect can be seen in smaller rooms, but is much less pronounced there so it is not normally noticed. Second, the noise reduction measured for a partition of a large room appears to vary depending on whether the speakers are in the corners or not, with lower noise reduction when the speakers are in the corners. This effect does not appear to depend on whether the speakers are faced into the corners or not. Data and any further observations and conclusions developed will be presented.

8:25

2aAA2. Evaluation of methods for isolating portable loudspeakers during sound transmission testing. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

When performing field sound transmission tests, the potential exists for structureborne flanking resulting from poor isolation between the loudspeakers and floor. Many of us resort to using furniture and other creative means of isolation with unpredictable results. This paper will present analysis of a handful mocked up isolation systems with the hope of finding a practical, portable solution.

8:45

2aAA3. Change in Canada's national building code—Overview of new requirements and of projects supporting the change. Christoph Hoeller, Berndt Zeitler, and Jeffrey Mahn (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The proposed 2015 edition of the National Building Code of Canada sees a major change in sound insulation requirements. Instead of prescribing requirements for the separating assembly only (in terms of STC values), the Code now sets requirements for the sound insulation performance of the complete system (in terms of Apparent Sound Transmission Class (ASTC) values), including flanking sound transmission. The National Research Council Canada is actively supporting the change in the Code by conducting various projects with industry associations from different construction sectors, in order to provide tools, guidance, and the necessary data for compliance. This presentation provides an overview of the new requirements and of the different paths to compliance. Furthermore, various projects conducted at the National Research Council Canada to support the Code change are presented, including tools and guidance to help practitioners. Detailed descriptions of two of the projects are given in two complementary presentations.

9:05

2aAA4. Change in Canada's national building code—Assessing flanking sound transmission in steel-framed constructions. Christoph Hoeller and Berndt Zeitler (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The proposed 2015 edition of the National Building Code of Canada sees a major change in sound insulation requirements. Instead of prescribing requirements for the separating assembly only (in terms of STC values), the Code now sets requirements for the sound insulation performance of the complete system (in terms of Apparent Sound Transmission Class (ASTC) values), including flanking

sound transmission. The National Research Council Canada is actively supporting the change in the Code by conducting various projects with industry associations from different construction sectors, in order to provide tools, guidance, and the necessary data for compliance. This presentation focuses on an ongoing joint project between the National Research Council Canada and the Canadian Sheet Steel Building Institute. In the project, the direct and flanking sound transmission in steel-framed assemblies are being investigated. In the presentation, an overview of the project is given, and updates on the current status of the investigation are provided, including measured data concerning the flanking sound transmission in steel-framed constructions.

9:25

2aAA5. Change in Canada's national building code—Assessing flanking sound transmission in concrete-masonry constructions. Berndt Zeitler, Frances King, Jeffrey Mahn, and Christoph Hoeller (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc-cnrc.gc.ca)

The proposed 2015 edition of the National Building Code of Canada sees a major change in sound insulation requirements. Instead of prescribing requirements for the separating assembly only (in terms of STC values), the Code now sets requirements for the sound insulation performance of the complete system (in terms of Apparent Sound Transmission Class (ASTC) values), including flanking sound transmission. The National Research Council Canada is actively supporting the change in the Code by conducting various projects with industry associations from different construction sectors, in order to provide tools, guidance, and the necessary data for compliance. This presentation focuses on a joint project between the National Research Council Canada and the Canadian Concrete Masonry Producers Association. In the project, the direct and flanking sound transmission in concrete masonry and hybrid building systems were investigated. For masonry walls in combination with concrete floors, the ASTC values were calculated according to ISO 15712-1. For masonry walls in combination with wood joist floors, the ASTC values were measured according to ISO 10848. Furthermore, the effect of linings on concrete masonry walls was investigated. This presentation will provide an overview of each of these issues, including results and recommendations.

9:45–10:00 Break

10:00

2aAA6. A new metric for evaluating mid- and high-frequency impact noise. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Impact isolation within multi-family dwellings is currently evaluated using the single number laboratory metric Impact Insulation Class (IIC) and associated field test metrics. There is wide acceptance that IIC does not adequately quantify low frequency impact noise such as thudding from footfalls which is prevalent in lightweight joist-framed construction. However, it is often assumed that mid- and high-frequency impact sources, such as heel clicks, dragging furniture, and dropping objects, are adequately characterized by IIC. Previous research by the authors have indicated that IIC does not adequately distinguish or rank-order between the acoustical performances of resilient matting located in the upper room of a floor ceiling assembly [LoVerde and Dong, J. Acoust. Soc. Am. 120, 3206 (2006); LoVerde and Dong, Proceedings of ICVS14 (2007)]. Many condominiums have regulations that require a minimum impact sound rating when replacing or installing hard surface finish flooring, and may require field testing to show compliance with the regulations. As expected, a field IIC metric like AIIC, NISR, or ISR is not a suitable descriptor for acoustical performance. A modified metric is defined that more accurately rank-orders the mid- and high-frequency impact noise performance of assemblies and is better suited for these performance requirements.

10:20

2aAA7. Auralization of sound insulation in buildings. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

In various surveys, it was found that people living in multi-family dwellings and apartment houses are annoyed by noise of their neighbors. Also, it seems that building regulations, for example, the German standard DIN 4109 "Sound insulation in buildings," are insufficient. The degree of annoyance is influenced by the personal conditions of the habitants (stress), the value of the dwelling and the duration the habitants live there. The effects on humans include disturbance of conversation or listening to the TV or radio in private dwellings as well as communication in office premises, reduced power of concentration during physical or mental work, and disturbance of sleep. All this strongly depends on the kind of noise signal (speech, music, footfall, etc.) and on the context, and thus, it is highly doubtful if single-number quantities such as the STL sufficiently describe the real situation. In this paper, a technique is presented for auralization of complex virtual acoustic scenes including airborne and structure-borne noise in buildings with particular respect to sound propagation across or between coupled rooms. Based on SEA-like sound propagation models in standardized prediction methods (EN 12354), algorithms are designed for FIR filtering of audio signals and applied in listening tests and for to creation of audio demos. The auralized sounds can be used during building design processes, in studies of human noise perception, and in development of new metrics for future building codes.

10:40

2aAA8. Measuring noise level reduction using an artificial noise source. Rene Robert, Kenneth Cunefare (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Office 002, Atlanta, GA 30332, rrobert6@gatech.edu), Erica Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Javier Irizarry (School of Bldg. Construction, Georgia Inst. of Technol., Atlanta, GA)

Residences near airports may be subjected to significant noise levels. The impact of aircraft traffic noise imposes a higher level of design consideration for the outdoor-to-indoor transmission of sound for residences. Noise Level Reduction (NLR) is a common metric used to quantify the ability for a building element to reduce the transmission of external sound pressure levels generated by aircraft. The aircraft noise mitigation measure is determined by the estimate of NLR and the building location in the airport's noise footprint. While

NLR may be measured using an actual traffic source (i.e., aircraft fly-overs), another practice is to perform measurements using a loudspeaker. An investigation is underway to better understand the loudspeaker methods of measuring NLR for buildings. Specifically, the study was tasked with quantifying various factors of these measurements such as angular dependency. NLR measurements were taken on the façade of a "test house" that was constructed for the purpose of the research. Although the "test house" is a single-room structure, the same procedures can be applied to buildings such as single family residences and multifamily dwellings, among others. The results of the analysis should provide a more comprehensive understanding of NLR measurement procedures implemented in sound insulation programs.

11:00

2aAA9. Challenges facing fitness center designers in multifamily buildings. Scott Harvey (Phoenix Noise & Vib., 5216 Chairmans Court, Ste. 107, Frederick, MD 21703, sharvey@phoenixnv.com)

Amenity spaces in multifamily and mixed use developments have become extremely popular and possibly mandatory to the economic success of the project. Of these amenity spaces fitness centers are extremely common and pose significant design challenges to the noise control engineer. This paper will compare several mitigation techniques used to control fitness center noise from today' prominent sources including treadmills, group exercise, weight machines, free weights, and cross fit weight drops. Mitigation in both wood and concrete structures will be addressed.

11:20

2aAA10. Sound isolation design options for mixed-use buildings. Sean Connolly (Big Sky Acoust., LLC, PO Box 27, Helena, MT 59624, sean@bigskyacoustics.com)

During the design of mixed-use buildings, developers typically have a general idea about the types of commercial tenants that will be in the building. However, those ideas can change after the building has been designed or built, and can range from an office space to retail to a fitness center to a restaurant with live music. This presents a challenge for noise control design to limit noise in residences located above the commercial spaces. Although some noise mitigation measures can be included as part of tenant improvements, the result may be limited by the building base structure decided upon when some commercial uses had not been originally considered. This paper discusses a menu of noise control design options for mixed-use buildings to separate commercial and residential spaces so a developer can make informed decisions about what types of commercial tenants to allow, core and shell constructions, costs of potential tenant improvements, and which options provide the most flexibility.

11:40

2aAA11. Comparisons of impact and airborne sound isolation among cross laminated timber, heavy-timber mill buildings, concrete, and light weight wood frame floor/ceiling assemblies. Matthew V. Golden (Pliteq, 616 4th St., NE, WA, DC 20002, mgolden@pliteq.com)

While laboratory measurements of the Impact Sound Pressure Level (ISPL) and Transmission Loss (TL) of concrete and lightweight wood-frame constructions are well understood, not much laboratory research has been conducted into the acoustical performance of CLT and heavy-timber mill buildings. Recent work on the performance of these wood based floor/ceilings has been presented at other conferences. This paper will review that previously published research for both the bare assemblies and assemblies that include resilient elements. These resilient elements include recycled rubber floor underlayment and sound isolation clip systems. This paper will then compare the performance of these assemblies to each other and to more common concrete and lightweight wood frame floor/ceiling assemblies. The analysis will also include the comparable strengths and weaknesses of each structural system along with the effectiveness of the various acoustical isolation techniques and their effectiveness on each of various floor ceiling assembly. It will be shown that the acoustical isolation techniques will perform differently on the various floor/ceiling assemblies.

Session 2aAB

Animal Bioacoustics: Bioacoustics Across Disciplines: Detecting and Analyzing Sounds

Elizabeth T. Küsel, Chair
Portland State University, Suite 160, 1900 SW 4th Ave., Portland, OR 97201

Contributed Papers

9:00

2aAB1. The effects of aging on detection of ultrasonic calls by adult CBA/CaJ mice. Anastasiya Kobrina and Micheal Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

Mice are frequently used as an animal model for human hearing research, yet their hearing capabilities have not been fully explored. Previous studies (Henry, 2004; Radziwon *et al*., 2009) have established auditory threshold sensitivities for pure tone stimuli in CBA/CaJ mice using ABR and behavioral methodologies. Yet, little is known about how they perceive their own ultrasonic vocalizations (USVs), and nothing is known about how aging influences this perception. The aim of the present study was to establish auditory threshold sensitivity for several types of USVs, as well as to track these thresholds across the mouse's lifespan. In order to determine how well mice perceive these complex communication stimuli, several CBA/CaJ mice were trained and tested at various ages in a detection task using operant conditioning procedures. Results showed that mice were able to detect USVs well into their lifespan, and that thresholds differed across USV types. Male mice showed higher thresholds for certain USVs later in life than females. In conclusion, the results suggest that mice are sensitive to their complex vocalizations even into old age, highlighting their likely importance for survival and communication.

9:15

2aAB2. Temporary threshold shift not found in ice seals exposed to single airgun impulses. Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, 1, 100 Shaffer Rd., Santa Cruz, CA 95060, coll@ucsc.edu), Brandon L. Southall (Southall Environ. Assoc. (SEA) Inc., Aptos, CA), Asila Ghoul, Andrew Rouse (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA), and Jillian M. Sills (Dept. of Ocean Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

We measured low-frequency (100 Hz) hearing thresholds in trained spotted seals (*Phoca largha*) and ringed seals (*Pusa hispida*) before and immediately after controlled exposures to impulsive noise from a small (10 in^3) seismic airgun. Threshold shifts were determined from psychoacoustic data, and behavioral responses to the impulse noise were scored from video recordings. Four incremental exposure conditions were established by manipulating both the distance and the operating pressure of the airgun, with received sound levels ranging from 190 to 207 dB $re 1\mu$Pa peak SPL and 165-181 dB re $1\mu Pa^2$-s SEL. We found no evidence of temporary threshold shift (TTS, $\geq$ 6 dB) in four subjects tested up to eight times each per exposure condition, including at levels previously predicted to cause TTS. Relatively low-magnitude behavioral responses were observed during noise exposure and indicate that individuals can learn to tolerate loud, impulsive sounds, but this does not necessarily imply that similar sounds would not elicit stronger behavioral responses in wild seals. The maximum exposure values used here can improve precautionary estimates for TTS onset from impulse noise in pinnipeds. However, additional studies using multiple impulses and/or higher exposures are needed to identify the actual noise conditions that induce changes in hearing sensitivity.

9:30

2aAB3. Classification of beaked whale and dolphin clicks measured by environmental acoustic recording system buoys in the northern Gulf of Mexico. Natalia A. Sidorovskaia, Kun Li (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA 70504, nas@louisiana.edu), Azmy Ackleh, Tingting Tang (Mathematics, Univ. of Louisiana at Lafayette, Lafayette, LA), Christopher O. Tiemann (R2Sonic LLC, Austin, TX), Juliette W. Ioup, and George E. Ioup (Physics, Univ. of New Orleans, New Orleans, LA)

The Littoral Acoustic Demonstration Center (LADC) has used its Environmental Acoustic Recording System (EARS) buoys to record sperm and beaked whales and dolphins, with frequencies up to 96 kHz, in the northern Gulf of Mexico in 2007 and 2010. The 2007 experiment was the first to record beaked whales in the Gulf. It has been found that there is considerable overlap in the band of beaked whale signals from 20 to 60 kHz with deepwater dolphin clicks, so traditional energy-band detectors have a high occurrence of false positives. Although acoustic measurements in this frequency range validated by visual observations have been limited, for the Gulf of Mexico species, progress is being made in automatically delineating clicks that belong to beaked whale species observed in the Gulf and those originating from dolphins. Spectrograms of the classified clicks are shown and compared to known spectrograms for beaked whale and dolphin species. Many of the spectrograms show an upsweep in the observed spectrum but others do not. Improved classifiers can provide higher accuracy estimates of the regional abundance trends and effects of environmental changes on both beaked whale and dolphin groups. [Research supported by BP/GOMRI, SPAWAR, ONR, NSF, and Greenpeace.]

9:45

2aAB4. Application of density estimation methods to datasets collected from a glider. Elizabeth T. Küsel, Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu), David K. Mellinger, and Sara L. Heimlich (Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Ocean gliders can provide an inexpensive alternative for marine mammal population density studies. Gliders can monitor bigger spatial areas than fixed passive acoustic recorders. It is a low-noise, low-speed platform, easy to set up, maneuver, and transport on land, deploy, and recover. They can be deployed for long periods and report near real-time results through Iridium modem. Furthermore, gliders can sense the environmental conditions of the survey area, which are important for estimating detection distances. The main objective of this work is to evaluate the use of ocean gliders for population density estimation. Current methodologies developed for fixed sensors will be extended to these platforms by employing both simulations and real experimental data. An opportunistic preliminary sea trial conducted in June 2014 allowed for testing of a Slocum glider fitted with an inexpensive acoustic recording system comprising of two hydrophones connected to an off-the-shelf voice recorder installed inside the glider. Acoustic data recorded in deep waters (>1500 m) off the western coast of Sardinia, Mediterranean Sea, showed the presence of sperm whale echolocation clicks. An improved experiment is planned for the summer 2015.

Preliminary results of both campaigns will be presented with an emphasis on population density estimation.

10:00

2aAB5. A technique for characterizing rhythms produced by singing whales. Eduardo Mercado (Dept. of Psychol., Univ. at Buffalo, Buffalo, NY 14260, emiii@buffalo.edu)

Structured sound sequences produced by baleen whales show strong rhythmicity. Such temporal regularity is widely acknowledged, but rarely analyzed. Researchers instead have focused heavily on describing progressive changes in sequential patterns of sounds revealed through spectrographic and aural impressions. Recent production-based analyses of humpback whale sounds suggest that the acoustic qualities of individual sounds can provide useful information about how whales are generating sounds and also may reveal constraints on cyclical production of sounds that help determine rhythmic patterns. Because past analyses have largely ignored the temporal dynamics of sound production, the extent to which whales vary the rhythmicity of sound production over time is essentially unknown. Production-based analyses can be combined with automated measures of temporal patterns of sound production to generate spectrogram-like images that directly reveal rhythmic variability within sound sequences. Rhythm spectrograms can reveal long-term regularities in the temporal dynamics of sound sequences that may provide new insights into how whales produce sequences as well as how they use them.

10:15–10:30 Break

10:30

2aAB6. An approximate model for foliage echoes to study biosonar function in natural environments. Chen Ming, Anupam K. Gupta, and Rolf Müller (Mech. Eng., Virginia Tech, 210 ICTAS II, 1075 Life Science Circle, Blacksburg, VA 24061, cming@vt.edu)

Natural environments are difficult for current engineered sonars but apparently easy for at least some species of echolocating bats. To better understand the information that foliage echoes provide for biosonar-based sensing, an approximate model of foliage echoes has been developed. The model simplifies the scattering problem through two key assumptions: (i) multi-path scattering was neglected and (ii) all leaves were assumed to be circular disks. Due to the latter, the parameters of the model were reduced to the number of the disks, their positions, sizes, and orientations. The exact far-field scatter from a disk (i.e., simplified leaf) in response to planar incident waves can be obtained by summation over an infinite series of spheroidal wave functions. To reduce the calculation time, the scattered field has been approximated by exponential, polynomial, and cosine fitting functions that also depend on the disk parameters. This allows the simulation of echoes from 100 leaves in 20 s on a standard PC. The model was able to reproduce the echo waveforms from dense and sparse foliages qualitatively. The model should thus be well suited for generation of large echo datasets to explore the existence and utilization of statistical invariants in the echoes from natural environments.

10:45

2aAB7. An examination of the biosonar problem associated with sperm whales foraging on jumbo squid in the Gulf of California. Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu), Kelly J. Benoit-Bird (College of Earth, Ocean, and Atmospheric Sci., Oregon State Univ., Corvallis, OR), William F. Gilly (Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), and Bruce Mate (Hatfield Marine Sci. Ctr., Oregon State Univ., Newport, CA)

The backscatter properties of jumbo or Humboldt squid (*Dosidicus gigas*) were examined *in-situ* by projecting simulated sperm whale (*Physter macrocephalus*) clicks to tethered squids. The incident signal was a broadband click with a peaked frequency of approximately 17 kHz. Echoes were collect for three different aspect angles; (broadside, anterior, and posterior, and for different body parts for one squid. The beak, eyes, and arms, probably via the sucker rings, played a role in acoustic scattering though their effects were small. An unexpected source of scattering was the cranium of the squid which provided a target strength nearly as high as that of the entire squid though the mechanism remains unclear. The data support the hypothesis that the pen may be an additional important source of squid acoustic scattering. The detection range of these squids for sperm whales were estimated by performing a parametric analysis. Although many of the squid migrate with the mesopelagic layer towards the surface at night, sperm whales have been observed to forage at depth throughout the day and night which would maintain a relative low echo to clutter ratio, increasing their detection range.

11:00

2aAB8. A comparative study of Pinna motions in horseshoe bats and old world leaf-nosed bats. Xiaoyan Yin (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Shanda South Rd. 27, Jinan, Shandong 250100, China, xiaoyanyin4@mail.sdu.cn), Phat Nguyen, Thomas J. Tucker (School of Visual Arts, Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (Rhinolophidae) and Old World leaf-nosed bats (Hipposideridae) are two closely related bat families that stand out for the dynamics of their ears during biosonar behaviors. In bats belonging to both families, the outer ears (pinnae) can undergo substantial, fast, and non-rigid shape changes while the animals emit their biosonar pulse trains. So far, characterization of these motions has been limited to either very general measures (e.g., of overall ear rotation) and/or limited data sets covering only a single species. Here, we have combined high-speed stereo vision with digital animation techniques to reconstruct and compare the motions of the pinnae and the head in one rhinolophid (*Rhinolophus ferrumequinum*) and two hipposiderid species (*Hipposideros armiger* and *Hipposideros pratti*). In parallel, we have also recorded the pulses and echoes received by the animals. We found that the pinna motions in both families frequently overlap in time with the arrival of the echoes, so they could have a functional relevance for echo reception. The pinna motions were found to follow similar patterns in all three species and could be decomposed into three main components. Beyond these fundamental similarities, there were also pronounced quantitative differences between the motions seen in the two families.

11:15

2aAB9. Single-sensor density estimation of highly broadband marine mammal calls. Elizabeth T. Küsel, Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu), and David K. Mellinger (Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Odontocete echolocation clicks have been used as a preferred cue for density estimation studies from single-sensor data sets. Such sounds are broadband in nature, with 10-dB bandwidths of 20 to 40 kHz or more. Estimating their detection probability is one of the main requirements of density estimation studies. For single-sensor data, detection probability is estimated using the sonar equation to simulate received signal-to-noise ratio of thousands of click realizations. A major problem with such an approach is that the passive sonar equation is a continuous-wave (CW) analysis tool (single-frequencies). Using CW analysis with a click's center frequency while disregarding its bandwidth has been shown to introduce bias to detection probabilities and hence to population estimates. In this study, the methodology used to estimate detection probabilities is re-evaluated, and the bias in sonar equation density estimates is quantified by using a synthetic data set. A new approach based on the calculation of arrivals and subsequent convolution with a click source function is also presented. Application of the new approach to the synthetic data set showed accurate results. Further complexities of density estimation studies are illustrated with a data set containing highly broadband false killer whale (*Pseudorca crassidens*) clicks. [Work supported by ONR.]

2a TUE. AM

11:30

2aAB10. Baleen whale localization using a dual-line towed hydrophone array during seismic reflection surveys. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington, 1501 NE Boat St., Seattle, WA 98195, shimah@ldeo.columbia.edu), Maya Tolstoy (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY), and William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA)

Three dimensional seismic reflection surveys use multiple towed hydrophone arrays for imaging the structure beneath the seafloor. Since most of the energy from seismic reflection surveys is low frequency, their impact on Baleen whales may be particularly significant. To better mitigate against this potential impact, safety radii are established based on the criteria defined by the National Marine Fisheries Service. Marine mammal observers use visual and acoustic techniques to monitor safety radii during each experiment. However, additional acoustic monitoring, in particular locating marine mammals, could demonstrate the effectiveness of the observations, and improve knowledge of animal responses to seismic experiments. In a previous study (Abadi *et al.*, 2014), data from a single towed seismic array was used to locate Baleen whales during a seismic survey. Here, this method is expanded to a pair of towed arrays and the locations are compared with an alternative method. The experimental data utilized in this presentation are from the seismic experiment conducted by the R/V Marcus G. Langseth near Alaska in summer 2011. Results from both the simulation and experiment are shown and data from the marine mammal observers conducted simultaneously with the experiment are used to verify the analysis. [Sponsored by NSF.]

TUESDAY MORNING, 3 NOVEMBER 2015 — RIVER TERRACE 2, 8:15 A.M. TO 11:15 A.M.

Session 2aAO

Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics: Passive-Acoustic Inversion Using Sources of Opportunity I

Karim G. Sabra, Cochair
Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, Fairfax, VA 22030

Chair's Introduction—8:15

Invited Papers

8:20

2aAO1. Passive acoustic remote sensing in a coastal ocean. Oleg A. Godin (Physical Sci. Div., NOAA-Earth System Research Lab, 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov) and Michael G. Brown (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL)

Sound propagation in shallow water over ranges large compared to ocean depth involves multiple reflections from the sea surface and seafloor. With the acoustic propagation environment being much more dynamic then in a deep ocean, noise interferometry faces new challenges but can also provide additional insights into physical processes in a coastal ocean. This paper will review the results on passive acoustic characterization of the seafloor and the water column, including ocean currents, that were obtained using the ambient noise data collected in 2012–2103 in the Straits of Florida. [Work supported by NSF and ONR.]

8:45

2aAO2. Alternative measurements and processing for extracting seabed information from sea-surface noise correlations. Martin Siderius, Joel Paddock, Lanfranco Muzi (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, siderius@pdx.edu), and John Gebbie (Metron Sci. Solutions, Portland, OR)

In recent years, both theoretical and experimental results have shown that the noise generated at the sea-surface from wind and waves contains valuable information about the seabed. A vertical hydrophone array together with beamforming has been a particularly useful configuration for estimating seabed properties. By cross-correlating a vertically upward looking beam with a downward looking beam (the endfire directions), the bathymetry and seabed layering can be determined. However, there may be additional information about the seabed found by cross-correlating beams in directions away from vertical endfire. In this presentation, two new measurement and processing configurations will be considered: noise cross-correlation of beams from a vertical array in directions away from endfire and cross-correlation on a towed horizontal array. For the vertical array, data and modeling show the existence of strong beam correlations coming from a direction consistent with the seabed critical angle. The towed horizontal array configuration, if possible, would provide an alternative to the vertical array for seabed surveying using noise. Measurements from data collected at several sites along with modeling will be used to explain the results from these new measurement and processing configurations.

Contributed Papers

9:10

2aAO3. Spatial sampling of seabed properties using a glider equipped with a short hydrophone array. Peter L. Nielsen, Jiang Yong-Min (Res. Dept., NATO-STO CMRE, VS Bartolomeo 400, La Spezia 19126, Italy, Peter.Nielsen@cmre.nato.int), Martin Siderius, and Lanfranco Muzi (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

Passive acoustic remote sensing of seabed geophysical properties using naturally occurring ambient noise and mobile underwater platforms has the advantage of providing critical environmental parameters over wide areas for sonar performance predictions. However, although technological advances have provided the opportunity to implement acoustic payloads on mobile underwater vehicles the extent and complexity of the acoustic sensors are limited and can therefore only serve as a sensing platform to derive lower resolution seabed properties. During the NATO-STO CMRE sea trial GLISTEN'15, a glider is equipped with an eight-element rigid hydrophone array to estimate the seabed properties based on natural occurring ambient noise. The glider will operate along tracks where the geoacoustic properties and stratification of the seabed are known to vary significantly from historical data. The results from the discrete sampling of the estimated seabed properties are presented and compared to estimates from a short and longer bottom-moored vertical hydrophone array along the tracks. The latest development in synthetic array extension to improve the resolution of the inferred seabed properties are applied and evaluated by comparison of results between the different arrays. The impact of the acquired seabed characteristics on long range acoustic propagation is assessed.

9:25

2aAO4. Improved passive bottom-loss estimation below 10 kHz using arrays deployable on autonomous underwater vehicles. Lanfranco Muzi, Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu), and Peter L. Nielsen (NATO-STO Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

Accurate modeling of acoustic propagation in the ocean waveguide is important for SONAR performance prediction, and requires, among other things, characterizing the reflection properties of the bottom. Recent advances in the technology of autonomous underwater vehicles (AUV) make it possible to envision a survey tool for seabed characterization composed of a short array mounted on an AUV. The bottom power reflection coefficient (and the related reflection loss) can be estimated passively by beamforming the naturally occurring marine ambient-noise acoustic field recorded by a vertical line array of hydrophones. However, the reduced array lengths required by AUV deployment can hinder the process, due to the inherently poor angular resolution. In this paper, data from higher frequencies are used to estimate the noise spatial coherence function at a lower frequency for sensor spacing beyond the physical length of the array. This results in higher angular resolution of the bottom loss estimate, while exploiting the large bandwidth available to current acquisition systems more efficiently than beamforming does. The technique, rigorously justified for a halfspace bottom, proves to be effective also on more complex bottom types, both in simulation and on experimental data.

Invited Papers

9:40

2aAO5. Optimized extraction of coherent arrivals from ambient noise correlations in a rapidly fluctuating medium, with an application to passive acoustic tomography. Katherine F. Woolfe (Naval Res. Lab., 672 Brookline St. SW, Atlanta, Georgia 30310, katherine.woolfe@gmail.com), Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA)

Ambient noise correlations can be used to estimate Green's functions for passive monitoring purposes. However, this method traditionally relies on sufficient time-averaging of the noise-correlations to extract coherent arrivals (i.e., Green's function estimates), and is thus limited by rapid environmental fluctuations occurring on short time scales while the averaging takes place. For instance, based on extrapolating results from a previous study [Woolfe *et al.,* 2015], passive ocean monitoring across basin scales (i.e., between hydrophones separated by $\sim$1000 km) may require at least 10 weeks of averaging time to extract coherent arrivals; but such an averaging time would be too long to capture some aspects of the mesoscale variability of the ocean. To address this limitation, we will demonstrate with simulation and data that the use of a stochastic search algorithm to correct and track these rapid environmental fluctuations can reduce the required averaging time to extract coherent arrivals from noise correlations in a fluctuating medium. The algorithm optimizes the output of an objective function based on a matched filter that uses a known reference waveform to track a set of weak coherent arrivals buried in noise.

10:05–10:25 Break

10:25

2aAO6. Ambient-noise inversion in ocean geoacoustics and seismic-hazard assessment. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca), Sheri Molnar (Univ. of Br. Columbia, Vancouver, BC, Canada), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper presents methodologies and results for the estimation of seabed and soil-column geophysical properties based on Bayesian inversion of ambient ocean-acoustic and seismic noise, respectively. In both marine and terrestrial applications, beamforming is applied to array-based measurements to select the direction of arrival of ambient-noise energy. For the ocean-acoustic application, wave-generated surface noise is recorded at a vertical line array and beamformed to extract up- and down-going energy fluxes from which bottom loss vs. angle can be computed and inverted for geoacoustic profiles. Results from ambient-noise measurements at the Malta Plateau are presented and compared to controlled-source inversions and core measurements. The terrestrial seismic application is aimed at earthquake-hazard site assessment, which requires knowledge of the shear-wave velocity profile over the upper tens of meters of the soil column. In this case, urban seismic noise is recorded on a geophone array and beamformed to determine the dominant direction of arrival over short time windows, from which the Rayleigh-wave dispersion curve can be estimated and inverted. Two sites with differing geology are considered, and results are compared to invasive (borehole) measurements.

10:50

2aAO7. High-resolution imaging of the San Jacinto fault zone with a dense seismic array and local seismic noise. Philippe Roux, Albanne Lecointre, Ludovic Moreau, Michel Campillo (ISTerre, Univ. of Grenoble, 1381 rue de la Piscine, Grenoble 38041, France, philippe.roux@ujf-grenoble.fr), Yehuda Ben-Zion (Univ. Of South California, Los Angeles, CA), and Frank Vernon (Scripps Inst. of Oceanogr., San Diego, CA)

A highly-dense Nodal array with 1108 vertical (10 Hz) geophone was deployed around the San Jacinto fault zone for ~4 weeks in 2014 in ~600 m x 600 m box configuration (nominal instrument spacing 10–30 m) centered on the Clark branch of the fault zone south of Anza. The array continuously recorded local ambient noise from which cross-correlations between each station pair were extracted for imaging purpose between 1 Hz and 20 Hz. Using subarrays made of 25 sensors, double beamforming was applied to separate body waves from surface waves. Focusing solely on surface waves in a first step, dispersion curves for surface wave group velocities are obtained with unprecedented accuracy at each point of a 10-m spacing grid. The data inversion reveals depth- and lateral-variations of local structural properties within and around the San Jacinto fault zone.

TUESDAY MORNING, 3 NOVEMBER 2015 — CLEARWATER, 8:00 A.M. TO 11:30 A.M.

Session 2aBA

Biomedical Acoustics and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications I

Guillaume Haiat, Cochair
Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de Technologie Superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K, Canada

Invited Papers

8:00

2aBA1. Simulation of ultrasound wave propagation in heterogeneous materials. Michael J. Lowe, Anton Van Pamel, and Peter Huthwaite (Mech. Eng., Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

The simulation of ultrasound waves propagating through heterogeneous materials is useful to a range of applications, including Nondestructive Evaluation (NDE) and materials characterization. In NDE, the challenge is to detect defects within the volume of a component against a background of noise of scattering from the inhomogeneities. In materials characterization, the challenge is to measure properties such as stiffness, texture, and spatial distributions of the inhomogeneities. Until recently, simulation of wave propagation through realistic detailed volumetric representations of heterogeneous materials was infeasible because of the huge computational requirements. However, it has recently become possible to do these kinds of simulations. This talk will show Finite Element model simulations of wave propagation in polycrystalline metals that are representative of the high performance alloys used in electricity power plant components. Validations by examining wave speed, attenuation, and backscatter will be discussed, and example deployments of the models will be presented.

8:20

2aBA2. Semi-analytical methods for the simulation of the ultrasonic non destructive testing of complex materials. Sylvain Chatillon, Vincent Dorval, and Nicolas Leymarie (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur yvette 91191, France, sylvain.chatillon@cea.fr)

During the last decade, the role of the NDT simulation has been continuously increasing and diversifying in all the industrial sectors concerned by complex inspection methods. The UT simulation tools gathered in the CIVA software developed at CEA LIST include beam and defect echoes computations. The propagation of elastic waves is computed using an asymptotic paraxial ray approximation. In the case of some heterogeneous materials, such as polycrystalline structures and welds, it requires the definition of equivalent propagation properties for which several methods have been developed. In the case of fine-grained polycrystals, the propagation is first

computed for a homogeneous effective medium. Attenuation filters are applied afterwards and structural noise is modeled as the echoes of a random distribution of scatterers. Their parameters can be set empirically or calculated based on microstructural properties. Polycrystals with coarser grains are modeled using Voronoi diagrams. The propagation in the complex structure of an austenitic weld can be modeled by describing it as piecewise homogeneous, though the impedance contrast between neighboring homogeneous domains may cause inaccurate results. As a part of the MOSAICS (ANR) project, the ray tracing method was extended to smoothly inhomogeneous anisotropic media and applied to more realistic descriptions of welds.

8:40

2aBA3. Optimizing the ultrasonic imaging of metallic components with high microstructural noise. Yousif Humeida, Paul D. Wilcox, and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

Ultrasonic arrays are used extensively in a wide range of non-destructive evaluation applications. Many engineering structures are manufactured from polycrystalline metals that result in high levels of microstructural noise making their inspection extremely challenging. In this paper, an optimization framework that uses fast and efficient forward models to simulate the ultrasonic response for both defects and the grains in scattering media is presented. Crucially, these models include both single and multiple scattering effects, and the optimal inspection depends on which type of scattering dominates. For a particular material, simple experimental measurements are used to extract information, such as attenuation and grain scattering coefficients, which is then used to populate models of the microstructural scattering that can be used in an optimization process. As a demonstration, the detectability of small (0.3 mm) defects in copper, a material with high microstructural scatter, is investigated, and the optimal array size, pitch, array location, central frequency, and frequency bandwidth are found. The performance of the optimization system has been evaluated using experimental measurements of receiver operating characteristic (ROC) curves. For the chosen example, the optimal array configuration results in a probability of detection of 90% with a 1% false alarm rate.

9:00

2aBA4. Imaging of a fractal pattern with time and frequency domain topological derivative. Vincent Gibiat, Xavier Jacob (PHASE-UPS, Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31062 cedex 9, France, vincent.gibiat@univ-tlse3.fr), Samuel Rodriguez (I2M, Bordeaux Univ., Talence, France), and Perrine Sahuguet (PHASE-UPS, Toulouse Univ., Toulouse, France)

While fractal boundaries and their ability to describe irregularity have been intensively studied, only a few studies are available on wave propagation in a medium where a fractal or quasi fractal pattern is embedded. Acoustical propagation in 1D or 2D domains can be modeled using Time Domain Finite Differences or in Frequency domain with Finite element methods. The fractal object is then considered as a subwavelength set of scatterers, and the problem becomes a multiple scattering one leading to acoustic localization. So, as some important part of the energy remains trapped inside the fractal pattern, imaging such a medium becomes difficult and imaging such complex media with classical tools as B-scan or comparable methods is not sufficient. The resolution of the inverse problem of wave propagation can then be achieved with the help of the more efficient imaging methods related with Time Reversal. Using the concept of topological derivative as defined in Time Domain Topological Energy and Fast Topological Imaging Method, respectively, in Time domain and Frequency domain is powerful in that case. Examples in 1D and 2D will be presented including the image obtained for a sliced sponge.

9:20

2aBA5. A non-dispersive discontinuous Galerkin method for the simulation of acoustic scattering in complex media. Abderrahmane Bendali (INSA-Toulouse, Institut de Mathématiques de Toulouse UMR 5219 CNRS, Toulouse, France), Hélène Barucq, Julien Diaz (Inria Bordeaux Sud-Ouest, EPC Magique 3D, Université de Pau et des Pays de l'Adour, UMR CNRS 5132., Pau, France), MBarek Fares (Algo Team, Cerfacs, Toulouse, France), Vanessa Mattesi, and Sébastien Tordeux (Inria Bordeaux Sud-Ouest, EPC Magique 3D, Université de Pau et des Pays de l'Adour, UMR CNRS 5132., Département de Mathématiques, Université de Pau, Ave. de l'Université, 64000 Pau, Toulouse 64000, France, sebastien.tordeux@gmail.com)

In the context of time harmonic acoustic wave propagation, the Discontinuous Galerkin Finite Element Method (DG-FEM) and the Boundary Element Method (BEM) are nowadays classical numerical techniques. On one hand, the DG-FEM is really appropriate to deal with highly heterogeneous media. In comparison with continuous Finite Elements Method (FEM), this method is well adapted to direct solver since its connectivity diagram is significantly smaller than the one of classical FEM. However, it suffers of numerical pollution: the numerical wave does not propagate at the correct velocity. This can be very problematic when this method is used on very large computational domain or at high frequency. On the other hand, the BEM is one of the most efficient method to deal with homogeneous media, especially when accelerated by a multipole method or thanks to the Adaptive Cross Approximation. Moreover, this method is really less affected by numerical pollution. However, BEM is not adapted to heterogeneous media. In this talk, we would like to present a DG-FEM whose shape functions are defined thanks to a BEM. This new numerical discretization method benefits from the advantages of BEM and DG-FEM: low pollution effect, ability to deal with highly heterogeneous media. Numerous simulations will show the efficiency and the accuracy of the method on large domains.

9:40

2aBA6. Influence of material heterogeneity on the distortion of a focused ultrasonic beam. Joseph A. Turner and Andrea Arguelles (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

Recent research associated with elastic wave scattering in heterogeneous materials has shown the importance of the Wigner transform in the formulation of the scattering problem to quantify the transducer beam pattern within the sample. The four-fold Wigner distribution function describes the combined time-frequency, space-wave vector domains simultaneously. To date, this approach has been used successfully to examine the diffusely scattered energy for both pulse-echo and pitch-catch measurement configurations. However, the defocusing effect caused by the scattering has received much less attention than the overall scattering. Here, this problem is posed

using a similar approach as the diffuse scattering problem. The energy distribution is shown to be an expansion in terms of the order of the material heterogeneity. Results for the first-order correction from the homogeneous beam pattern are shown with respect to the various parameters of the problem including the frequency, transducer element size, the transducer focus, and the material heterogeneity (here defined in terms of the grain size of a polycrystalline sample). Experimental results of an ultrasonic pulse propagating through a heterogeneous layer will be shown and compared with the model.

10:00–10:15 Break

10:15

2aBA7. Ultrasonic characterization of cohesive and adhesive properties of adhesive bonds. Michel Castaings, Emmanuel Siryabe, Mathieu Renier, Anissa Meziane (I2M, Univ. of Bordeaux, 351 cours Liberation, I2M (A4) - Univ. of Bordeaux, Talence 33400, France, michel.castaings@u-bordeaux.fr), and Jocelyne Galy (IMP, INSA Lyon, Villeurbanne, France)

The increasing use of adhesively bonded joints requires non-destructive evaluation methods to be developed, mostly for safety reasons. An adhesive joint can be divided into two sensitive zones that may cause mechanical failure: the body of the adhesive layer (cohesive zone) and the interphase between that adhesive and one of the substrate (adhesion zone). Weaknesses of these cohesive or adhesive zones can come, for example, from an incomplete curing of the adhesive or from inappropriate, initial treatment of the substrate surface, respectively. The present research attempts to characterize mechanical properties, which are representative of the adhesive and cohesive states of adhesively bonded assemblies, using a through-transmission ultrasonic method. To simplify the approach, the assemblies are made of two aluminum substrates and an epoxy-based adhesive layer. Six samples have been manufactured with various levels of cohesion or adhesion. Inverse problems are then solved to infer the elastic *moduli* of the adhesive or stiffness coefficients, which are modeling the interfacial adhesion. The potential, limits, and outlook of the proposed method are discussed.

Contributed Paper

10:35

2aBA8. A fractional calculus approach to the propagation of waves in an unconsolidated granular medium. Vikash Pandey and Sverre Holm (Dept. of Informatics, Univ. of Oslo, Postboks 1080, Blindern, Oslo 0316, Norway, vikashp@ifi.uio.no)

Our study builds on the work of Buckingham [JASA (2000)] which employed a grain-shearing (GS) model to describe the propagation of elastic waves in saturated, unconsolidated granular materials. He ensemble averages the random stick-slip process which follows the velocity gradient set up by the wave. The stick-slip process is due to the presence of micro-asperities between the contact surfaces of the grains. This is a strain hardening process which is represented by a time-dependent coefficient in the Maxwell element, besides the coefficient also gives the order of the loss term in the wave equations. We find that the material impulse response derived from the GS model is similar to the power-law memory kernel of fractional calculus. The GS model then gives two equations; a fractional Kelvin-Voigt wave equation for the compressional wave and a fractional diffusion equation for the shear wave. These equations have already been analyzed extensively in the framework of fractional calculus. Since the Kelvin-Voigt model is used in biomechanics of living tissue, we believe the GS theory could offer insights into ultrasound and elastography as well. The overall goal is to understand the role of different material parameters which affect wave propagation.

Invited Papers

10:50

2aBA9. Improving robustness of damage imaging in model-based structural health monitoring of complex structures. Patrice Masson, Nicolas Quaegebeur, Pierre-Claude Ostiguy, and Peyman Y. Moghadam (GAUS, Mech. Eng. Dept., Universite de Sherbrooke, 2500 Blvd. Universite, Sherbrooke, QC J1K 2R1, Canada, Patrice.Masson@USherbrooke.ca)

Model-based Structural Health Monitoring (SHM) approaches offer higher resolution in damage detection and characterization. However, the performance of damage imaging algorithms relies on the quality of the model and the proper knowledge of its parameters, which might prove challenging to obtain for complex structures. In this paper, models are first compared for ultrasonic guided wave generation by bonded piezoceramic (PZT) transducers, from the well-known pin-force model to analytical approaches taking into account the detailed interfacial shear stress under the PZT, and including an electro-mechanical hybrid model. Then, the modeling of guided wave propagation in complex structures is investigated, more specifically in composite structures, considering: (1) dependency of phase velocity and damping on the angle, (2) steering effect due to the anisotropy of the structure, and (3) full transducer dynamics. Validation of the models is conducted on isotropic and composite materials, by comparing amplitude curves and time domain signals with simulation results from Finite Element Models and with experimental measurements using a 3D laser Doppler vibrometer for principal and non-principal directions. Finally, the sensitivity of the damage imaging algorithms to variability in the model parameters is studied, and the benefit of identifying those parameters *in-situ*, prior to damage imaging, is demonstrated.

11:10

2aBA10. Reconstructing complex thickness variations with guided wave tomography. Peter Huthwaite, Michael J. Lowe, and Peter Cawley (Mech. Eng., Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

Wall thickness mapping is very important for quantifying corrosion within the petrochemical industry. One approach is guided wave tomography, where Lamb-type waves, which travel at different speeds depending on the thickness, due to dispersion, are passed through the region of interest. Wave speed is then reconstructed by a tomographic inversion approach and is converted to thickness by the known dispersion relationship. This approach relies on the assumption that guided waves scatter from the varying thicknesses within the plate in the same way that they would from the equivalent varying velocity field. This talk will investigate the accuracy of this assumption for the complex thickness variations associated with corrosion defects found in industry, and discuss potential approaches to mitigate the effects of any errors associated with this.

TUESDAY MORNING, 3 NOVEMBER 2015 ORLANDO, 8:00 A.M. TO 9:40 A.M.

Session 2aEAa

Engineering Acoustics: Vector Sensors: Theory and Applications

Michael V. Scanlon, Chair

RDRL-SES-P, Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

8:00

2aEAa1. Using vector sensors to measure the complex acoustic intensity field. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington and Appl. Phys. Lab., Seattle, WA)

The acoustic intensity vector field, defined as the product of pressure particle velocity fields, describes the flow of energy from an acoustic source. Following a brief introduction of acoustic vector sensors, which includes some direct measurements of acoustic intensity, the intensity field is shown to be composed of active and reactive components. Active intensity streamlines depict the time-averaged flow of acoustic energy and reveal characteristics of acoustic propagation including environmental influences. These streamlines do not characterize reactive intensity, which corresponds to the portion of acoustic intensity that time-averages to zero. Reactive intensity is significant in the near-field of a source and in environments where multipath interfere occurs. To examine the interplay between active and reactive acoustic intensity, the acoustic field generated by an airborne source positioned well above a water surface is presented. Acoustic measurements of a passing airplanes, made simultaneously above and below this sea-surface, are used to demonstrate properties of active and reactive intensity, including how reactive intensity can serve as an indicator of source altitude and range.

8:20

2aEAa2. Acoustic vector sensors: Principles, applications, and practical experience. Miriam Haege (Sensor Data and Information Fusion, Fraunhofer Inst. for Commun., Information Processing and Ergonomics, Fraunhoferstrasse 20, Wachtberg 53343, Germany, miriam.haege@fkie.fraunhofer.de)

Acoustic sensors can be applied in both the civilian and military domain in order to detect risky acoustic sources and localize them. In the case of military use, they are of high importance in sniper and gunshot detection as well as in the classification of firearms. On the civil side, acoustic sensors are employed, e.g., in environmental acoustic monitoring. This paper focuses on the usage of a special class of acoustic sensors, the so called acoustic vector sensors. Such sensors measure the scalar acoustic pressure field as well as the vectorial velocity field at a single location. The construction and the functional principle of an acoustic vector sensor will be described. This type of sensor was applied in a series of field trials, e.g., sniper localization and aircraft detection. The paper presents the experimental results received from the corresponding measurements and discusses the experiences gained with this sensor type.

8:40

2aEAa3. Acoustic particle velocity sensor: Application to infrasonic detection. Latasha Solomon (US Army Res. Lab, 2800 Powder Mill RD, Adelphi, MD 20783, latasha.i.solomon.civ@mail.mil)

Infrasonics is the study of low-frequency acoustics that is below the normal limit of human hearing. Infrasound has the ability to propagate extremely long ranges and is often utilized to monitor naturally occurring phenomena and explosives. Acoustic particle velocity sensors have shown promise in detection and localization of transient signals in the audio range such as small arms fire, mortars, and rocket propelled grenades. Ideally, this sensor can be used to detect various targets spanning a broad range of frequencies to include that of infrasound. The primary objective of this research is to characterize the acoustic vector sensor's localization performance for infrasonic sources given varying atmospheric conditions using algorithms developed at the Army Research Laboratory (ARL).

8:55

2aEAa4. Contributions to the intensity field in shallow water waveguides. Geoffrey R. Moss, Thomas Deal (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841, geoffrey.moss@navy.mil), and Kevin B. Smith (Naval Postgrad. School, Monterey, CA)

In a typical waveguide propagation, the acoustic intensity field is made up from contributions by a direct wave, bottom and surface reflected waves, and a number of interface waves. In shallow water environments, the laterally traveling headwave, generated by interaction with a faster propagating bottom layer, may become important. Several numerical techniques are used to calculate intensity fields in shallow water environments including normal modes, parabolic equation, and finite element methodologies. Bottom interactions are modeled with equivalent fluid properties, and the relative influence of the laterally traveling head wave is examined for several bathymetries of interest. Each codes' solution and merits are compared when calculating both propagating (active) and stationary (reactive) low frequency acoustic intensities.

9:10

2aEAa5. High sensitive MEMS directional sound sensor with comb finger capacitor electronic readout. Daniel Wilmott, Fabio Alves, and Gamani Karunasiri (Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, gkarunas@nps.edu)

The conventional directional sound sensing systems employ an array of spatially separated microphones to achieve directional sensing by monitoring the arrival times and amplitudes of different microphones. However, there are insects such as Ormia ochracea fly that can determine the direction of sound using its miniature-hearing organ much smaller than the wavelength of sound it detects. The fly's eardrums that are coupled mechanically with separation merely by about 1 mm have remarkable sensitivity to the direction of sound. Our MEMS based sensor, which consists of two 1 mm^2 wings connected in the middle, similar to the fly's hearing system, was designed and fabricated using silicon on insulator (SOI) substrate. The vibration of the wings in response to incident sound at the bending resonance was measured using a laser vibrometer found to be about 1 μm/Pa. For measuring sensor response electronically, comb finger capacitors were integrated on to the wins and the measured output using a MS3110 capacitive to voltage converter was found to be about 25 V/Pa. The fabricated sensors showed $\cos^2 q$ directional response similar to a pressure gradient microphone. The directional response of the sensor was measured down to about 30 dB.

9:25

2aEAa6. Cantilever-based acoustic velocity sensors. Joseph A. Bucaro (Excet Inc., 4555 Overlook Ave. SW, Naval Res. Lab., WA, DC 20375, joseph.bucaro.ctr@nrl.navy.mil), Nicholas Lagakos (Sotera Defense Solutions, Mclean, VA), Brian H. Houston, and Maxim Zalalutdinov (Naval Res. Lab., WA, DC)

This paper discusses progress made on the design of an acoustic velocity sensor. An analytic model was developed for the frequency response of a slender cantilever rod forced by the pressure gradient and particle velocity associated with an acoustic wave propagating in a fluid. The model, validated with acoustic response measurements in air, was used to design cantilever sensors, which respond predominately to acoustic particle velocity. One such design utilizes a short cantilever formed from a 125 çm silica glass fiber immersed in a viscous fill fluid whose lateral tip displacement is detected using a multi-fiber optical probe. This velocity sensor is predicted to be able to detect fairly low acoustic sound levels in water. Progress has been made in instrumenting a large pool at NRL to allow accurate propagating acoustic wave response measurements in water of these new velocity sensors down to frequencies below 5 Hz. Measurements made in this facility on various cantilever sensors will be presented and discussed. [Work supported by ONR.]

Session 2aEAb

Engineering Acoustics: Analysis of Sound Sources

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

10:15

2aEAb1. Numerical and experimental study of acoustic horn. Clebe J. Vitorino (Pontifical Catholic Univ. of Parana – PUCPR, Curitiba, Paraná, Brazil), Nilson Barbieri (Universidade Tecnológica Federal do Paraná - UTFPR, Curitiba, Paraná, Brazil), Key F. Lima (Pontifical Catholic Univ. of Parana – PUCPR, Rua Imaculada Conceiçao, 1155, Curitiba, Paraná 80215-901, Brazil, key.lima@pucpr.br), and Renato Barbieri (Univ. of the State of Santa Catarina, Joinville, Santa Catarina, Brazil)

Horns are acoustic elements specially designed for maximum transmission of sound pressure and they are used, for example, in sound systems (mainly, external and warning equipment), musical instruments, cleaning apparatus, receivers, and microwave transmitters. The objective of this work is to develop a methodology for obtaining the ideal shape of acoustic horns by comparing numerical data obtained by computational simulation of mathematical model (Finite Element Method—FEM) and experimental data obtained by acoustic tests in laboratory (two microphones method). The main steps to obtain the ideal geometry of acoustic horns are the definition of the objective function, the evaluation of this function and the optimization technique used. The reflection coefficient of the wave is the parameter optimized by the objective function by using the PSO (Particle Swarm Optimization) method. The results obtained for the optimization process were very satisfactory, especially for the correct control of the optimized geometry. The numerical and experimental data showed some differences due to limitations of the numerical model, but the results were good and appear promising.

10:30

2aEAb2. Sound amplification at T-junctions with merging mean flow. Mats Åbom and Lin Du (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Teknikringen 8, Stockholm 10044, Sweden, matsabom@kth.se)

This paper reports a numerical study on the aeroacoustic response of a rectangular T-junction with merging mean flow. The Mach number of the grazing flow in the main duct is fixed at 0.1. The primary motivation of the work is to explain the phenomenon of high level sound amplification, recently seen experimentally, when introducing a small merging bias flow. The acoustic results are found solving the compressible Linearized Navier-Stokes Equations (LNSEs) in the frequency domain, where the base flow is first obtained using RANS with a k-ε turbulence model. It is found that the base flow changes significantly with the presence of a small bias flow. Compared to pure grazing flow, a strong shear layer is created in the downstream main duct, starting from the T-junction trailing edge. That is, the main region of vortex-sound interaction is moved away from the junction to a downstream region much larger than the junction width. The flux of fluctuating enthalpy is calculated to estimate the acoustic source power due to the fluid sound interaction.

10:45

2aEAb3. Simulation of surge in the induction system of turbocharged internal combustion engines. Rick Dehner, Ahmet Selamet, and Emel Selamet (Mech. and Aerosp. Eng., The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, dehner.10@osu.edu)

A computational methodology has been developed to accurately predict the compression system surge instabilities within the induction system of turbocharged internal combustion engines by employing one-dimensional nonlinear gas dynamics. This capability was first developed for a compression system installed on a turbocharger flow stand, in order to isolate the surge physics from the airborne pulsations of engine. Findings from the turbocharger stand model were then utilized to create a separate model of a twin, parallel turbocharged engine. Extensive development was carried out to accurately characterize the wave dynamics behavior of induction system components in terms of transmission loss and flow losses for the individual compressor inlet and outlet ducts. The engine was instrumented to obtain time-resolved measurements for model validation under stable, full-load conditions and during surge instabilities. Simulation results from the turbocharger stand and engine agree well with the experimental data from their corresponding setups, in terms of both the amplitude and frequency of surge oscillations.

11:00

2aEAb4. Pressure ripple amplification within a hydraulic pressure energy harvester via Helmholtz resonator. Ellen Skow, Kenneth Cunefare, and Zachary Koontz (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, eskow3@gatech.edu)

Noise within a hydraulic system is a high-intensity ambient energy source that can be harvested to enable wireless sensor nodes. The noise is typically due to deterministic sources, generally caused by pumps and actuators, and has dominant frequency components around hundreds of Hertz. Hydraulic pressure energy harvesters (HPEH) are centimeter-sized devices that convert the noise into electricity via coupling of the fluid to piezoelectric materials. HPEH devices produce milliwatt level power, which is sufficient for low-energy sensor nodes. A common device used for amplifying or absorbing acoustic energy is a Helmholtz resonator (HR). Incorporation of a HR into an HPEH has been predicted to increase the HPEH power response by up to 7 dB. The properties of hydraulic oil cause HPEH-sized HR to resonate well above the dominant frequencies. Added compliance into the resonator allows the resonance to be tuned closer to the dominant frequency within the hydraulic system. A prototype HPEH with an integral Helmholtz resonator was developed and tested. The results are also compared to an electromechanical model developed for HPEH-HR devices.

Session 2aED

Education in Acoustics and Musical Acoustics: Effective and Engaging Teaching Methods in Acoustics

David T. Bradley, Cochair
Physics × Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604

Preston S. Wilson, Cochair
Mech. Eng., The University of Texas at Austin, 1 University Station, C2200, Austin, TX 78712

Invited Papers

8:00

2aED1. Finite element illustrations in the classroom. Uwe J. Hansen (Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu)

"You hear, you forget. You see, you remember. You do, you understand." This is Tom Rossing's favorite education quote. Solutions to wave equations generally result in traveling waves. Imposing boundary conditions usually limits these solutions to discrete normal modes. The one dimensional elastic string is an easy, accessible example, illustrated frequently with a long spring. The two dimensional example of a rectangular stiff plate is a little more complex, and the normal modes are often illustrated with Lissajous figures on a plate driven by a shaker. Whille full blown FEA programs are prohibitively expensive, ANSYS has an educational package available to students at nominal cost with sufficient memory to demonstrate normal mode vibrations in moderately complex structures. Normal mode vibrations in a rectangular plate with a number of different boundary conditions will be illustrated.

8:20

2aED2. Methodology for teaching synthetic aperture sonar theory and applications to undergraduate physics and oceanography majors. Murray S. Korman and Caitlin P. Mullen (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Undergraduate senior level physics majors taking Acoustics and oceanography majors taking Underwater Acoustics and Sonar learn about transmitting and receiving arrays (in one unit of their course) and do laboratory experiments to support and enhance the theoretical developments. However, there is a need to expose the students to a detailed unit on synthetic aperture sonar (SAS) while research at USNA is in progress and a teaching laboratory workstation is being developed. This paper communicates the teaching strategy on the topics describing: (a) how a strip-mapped SAS system works, (b) how matched-filtering relates to pulsed compression for a linearly modulated (LFM) pulsed chirp, (c) how synthetic aperture resolution is vastly improved over a conventional acoustic array, (d) how Fourier analysis is used in SAS, and (e) how a data set of N echoes can be used within a back-projection algorithm to obtain a two dimensional reflectivity image of an area (sea floor). Key points are (1) theory with visualizations to convey the teaching material to seniors in a two week period of time, (2) computer simulations, (3) classroom demonstrations in a water tank or in air, and (4) student involvement in a mini-research project using the computers and demonstration apparatus.

8:40

2aED3. Teaching the characterization of performance spaces through in-class measurements at Bates Recital Hall at the University of Texas at Austin. Michael R. Haberman, Kyle S. Spratt (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), Dan Hemme (BAi, LLC, Austin, TX), and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA)

The characterization of a performance space provides an excellent opportunity to provide students with first-hand experience with many fundamental aspects of room acoustics including reverberation, linear time-invariant systems, measurement methods, and post-processing of real-world data to estimate room metrics. This talk reports recent measurements made in Bates Recital Hall at the University of Texas at Austin (UT) as part of the graduate course on architectural acoustics in the acoustics program at UT. For one class period, students participated in acoustical measurements in the 700-seat venue. The measurements consisted of recording the signal at numerous locations within the room resulting from various on-stage excitations including exponential chirps, interrupted pink noise, and balloon-pops. The audio files captured during the experiments were provided to the students for calculation of the impulse response at the measurement positions and associated room metrics such as reverberation time, bass ratio, clarity index, and initial time-delay gap. Learning outcomes from this approach will be discussed in light of the experiential learning model which emphasizes abstract conceptualization, experimentation and experience, and reflective observation.

9:00

2aED4. Fundamentals of acoustics, vibration and sound, underwater acoustics and electroacoustic transduction at UMass Dartmouth. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This paper summarizes the teaching methods and material covered in four introductory graduate classes that are jointly offered as senior technical electives in Electrical Engineering, Mechanical Engineering, and Physics. The possibility of teaching two or three slightly different courses can be done with common lectures by designing different homework, class projects, and examinations for the mechanical engineering and electrical engineering students. Mechanical students are typically well prepared in mechanics and materials while find the use of equivalent electrical circuits more challenging. The reverse is typical for the electrical students. The classes involve many acoustic demonstrations and those are universally appreciated by students with all backgrounds. A number of examples will be presented.

9:20

2aED5. Standards-based assessment and reporting in introductory acoustics courses. Andrew C. Morrison (Joliet Junior College, 1215 Houbolt Rd., Natural Sci. Dept., Joliet, IL 60431, amorrison@jjc.edu)

Standards-based assessment and reporting (SBAR), also referred to as standards-based grading and other names, is a system for tracking student learning in a course and assigning a grade. SBAR replaces traditional grading systems by eschewing points-based assignments and evaluations. Instead, students are assessed over specific learning objectives called standards. SBAR has several advantages including: emphasis on topic mastery, development of intrinsic motivation for learning, and simplifying the grading process. I have implemented SBAR in an introductory acoustics course for music technology, sonography, and general education students. In order to change from a traditional grading system to SBAR several steps were taken. Standards were developed, assessments were written, and course policies were drafted. I will discuss the system as implemented, some of the challenges faced, and suggest ways in which the acoustics education community might provide support for others wanting to implement SBAR in their courses.

9:40

2aED6. Using sound and music to teach waves. Gordon Ramsey (Physics, Loyola Univ. Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

Sound and music are based on the properties of waves. They are also motivating topics for learning many subjects, from music to physics. The most recent "New Generation Science Standards" (NGSS) requires coverage of waves at all K-12 levels. Studies have shown that active student involvement is important in science education for helping the students understand physical concepts. These facts imply that music and acoustics are perfect avenues for teaching the concepts of waves. Even at the college level, non-science majors can understand how music and physics are related through the understanding of wave phenomena. There are many demonstrations, laboratory investigations and hands-on group activities that can be done at all levels. This paper suggests ways to incorporate sound and music to present waves at the levels of middle school, high school, and beginning college.

10:00–10:15 Break

10:15

2aED7. Characterization and design of sound absorber materials. Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., WA, DC 20064, diegoturo@gmail.com), Aldo A. Glean (Saint-Gobain Northboro R&D Ctr., CertainTeed Corp., Northboro, MA), Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, WA, DC), Teresa Ryan (Eng., East Carolina Univ., Greenville, NC), and John A. Judge (Mech. Eng., The Catholic Univ. of America, WA, DC)

Sound absorbing materials are widely used to mitigate noise in indoor environments. Foams and fiberglass are commonly used for passive noise control in the automotive, aerospace industries and for architectural design. The physics of sound absorption in porous materials is not typically included in introductory acoustics courses. However, characterization and design of sound absorbing materials and modeling of their properties can be valuable for students interested in applied acoustics. In this laboratory-oriented course at the Catholic University of America, we cover design of multilayered sound absorbers and experimental procedures for testing such materials. These include an introduction to data acquisition using LabVIEW and post-processing (with Matlab) of recorded sound as well as microphone calibration, and measurement of sound pressure level, and the frequency response function of a speaker. The second part of the course focuses on room acoustics, acoustic properties of materials, impedance tube measurements, modeling of sound propagation in porous media, and design of sound absorbers. We validate the Zwikker and Kosten model using a sample with straight cylindrical pores and apply the Delany-Bazley model to predict acoustic behavior of fibrous materials. Finally, multilayered materials are designed using the impedance translation theorem and tested with impedance tube measurements.

10:35

2aED8. Evolution of experiential learning in an acoustics elective course. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 49504, dludwigs@kettering.edu)

Starting in 2007, PHYS-388, Acoustics in the Human Environment, has been part of the Acoustics Minor at Kettering University. Originally envisioned to capture foundational concepts of acoustics that would be essential to a wide variety of engineering and scientific applications, this course is aimed at a junior/senior level audience to reflect the initiative and maturity required of the student. Topics emphasize the interdisciplinary nature of acoustics in industry, incorporating digital signal processing, psychoacoustics, and applications in room acoustics and environmental noise. Its evolution from a face-to-face studio environment to a hybrid, and then fully online course has retained a hands-on experiential course design and themes of art and design. Challenges arose in teaching the course online, and solutions to promote learner engagement and consistency are described.

10:55

2aED9. Physics of music field trips. Juliette W. Ioup and George E. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu)

Many studies on educational techniques have shown that students learn best when they are actively engaged. In introductory physics classes, this means each student makes measurements and performs calculations; in music performance classes, this means each student practices a musical instrument and analyzes music scores. In the two-semester sequence of Physics of Music lectures and laboratories taught at UNO, students participate in a variety of hands-on activities, including measuring quantities for available musical instruments and then calculating various physical parameters from the measured values. Another helpful activity is going on "field trips" to other locations, such as the auditorium in the UNO Performing Arts Center (once to study the concert grand piano there and once to make room acoustic measurements), the small recording studio on campus (lecture given there by the UNO recording engineer demonstrating various types of equipment), and to a chapel across the street from campus (pipe organ demonstrated by the organist of the chapel, acoustic measurements made by students). An extra benefit is that these trips are attractive to both non-science and science students, assisting in both recruitment and retention. Suggestions for and cautions about various field trips from experiences teaching these courses will be presented.

Contributed Papers

11:15

2aED10. Investigation of acoustical spectral analysis to gain a better understanding of tone development in single reed instruments. Charles E. Kinzer (Dept. of Music, Longwood Univ., Farmville, VA 23909, kinzerce@longwood.edu), Stanley A. Cheyne, and Walter C. McDermott (Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA)

A series of frequency spectra were measured and analyzed, and the results were used for comparison on a series of tone development exercises for saxophone and clarinet players. The musical exercises were focused on manipulations of the embouchure, oral cavity, and vocal tract as a means for altering the overtone content of the tone produced. The acoustical measurements were used to foster greater understanding on the part of the musicians of the importance of the player's physiology on the production of a musical tone, and help develop the player's ability to alter the tone in an intentional manner. The results of this study will be presented and discussed.

11:30

2aED11. The science of adhesion. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

This presentation is designed to promote the study of science to pre-college audiences. It has been well-received at a variety of pop culture conventions by attendees that vary from high school students interested in building better costumes to NASA Ph.D.'s whose mission is to build better satellites. ASA attendees will be strongly encouraged to use this (or a similar) presentation as an outreach tool to help recruit the next generation of STEM scholars.

11:45

2aED12. Development of an educational electro-mechanical model of the middle ear. Juliana Saba, Hussnain Ali, Jaewook Lee, John Hansen, Son Ta, Tuan Nguyen, and Cory Chilson (Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, jns109020@utdallas.edu)

In the United States, 1 in 5 people (20%, or 48.1 M), 12 years or older, have hearing loss in one or both ears,[1, 2] and approximately 3 out of 1000 children are born with some degree hearing loss.[1, 3]Educating the public on potential causes of hearing loss is often overlooked but imperative; like the remedial task of listening to headphones at a high volume. This in part is due to lack of interactive educational tools to demonstrate sound sensation/perception and apparatuses of the natural safety mechanisms against high intensity sounds. This study aids to encourage particularly a younger generation by increasing public health awareness with the design of a standalone, interactive, and educational electro-mechanical model that exhibits middle ear motion. The model includes: (i) anatomical 3-bone configuration (malleus, incus, and stapes), (ii) cochlea fluid environment, (iii) electrical stimulation of auditory nerve fibers, and (iv) an informational display regarding the natural safety mechanism, sound conduction process, the role of the cochlea in sound sensation, and how cochlear implants/hearing aids assist auditory rehabilitation. Since the model encourages hands-on learning, its placement is desired in either a classroom or a museum striving to reduce any ear-damaging habits and negligence by increasing cognizance.

Session 2aNS

Noise: Damage Risk Criteria for Noise Exposure I

Richard L. McKinley, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Hilary L. Gallagher, Cochair
Air Force Research Lab., 2610 Seventh St. Bldg. 441, Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—8:30

Invited Papers

8:35

2aNS1. Evaluation of high-level noise exposures for non-military personnel. William J. Murphy, Edward L. Zechmann, Chucri A. Kardous (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), and Scott E. Brueck (Div. of Surveillance Hazard Evaluations and Field Studies, Hazard Evaluations and Tech. Assistance Branch, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Noise-induced hearing loss is often attributed to exposure to high-level impulsive noise exposures from weapons. However, many workers in industries such as mining, construction, manufacturing, and services are exposed to high-level impulsive noise. For instance, construction workers use pneumatic tools such as framing nailers that can produce impulses at levels of 130 dB peak sound pressure level or greater. Miners have exposures to roof bolters and jack-leg drills that produce more of a continuous, but highly impulsive noise. In some areas of manufacturing the exposures include drop forge processes which can create impacts of more than 140 dB. Finally, in the services sector, law enforcement personnel who maintain proficiency with firearms experience the full gamut of small-caliber firearms during training. This paper will examine the noise exposures from recordings that the NIOSH Hearing Loss Prevention Team have collected. The noises will be evaluated with different damage risk criteria for continuous and impulse noise where appropriate.

8:55

2aNS2. Role of the kurtosis metric in evaluating hearing trauma from complex noise exposures—From animal experiments to human applications. Wei Qiu (Auditory Res. Lab., SUNY Plattsburgh, 101 BRd. St., Plattsburgh, NY 12901, wei.qiu@plattsburgh.edu), Meibian Zhang (Zhejiang Provincial Ctr. for Disease Control and Prevention, Hangzhou, China), and Roger Hamernik (Auditory Res. Lab., SUNY Plattsburgh, Plattsburgh, NY)

A number of animal experiments and epidemiologic studies in humans have demonstrated that current noise standard underestimated hearing trauma by complex noise. While energy and exposure duration are necessary metrics they are not sufficient to evaluate the hearing hazard from complex noise exposure. The temporal distribution of energy is an important factor in evaluating NIHL. Kurtosis incorporates in a single metric all the temporal variables known to affect hearing (i.e., peak, inter-peak interval, and transient duration histogram) that makes kurtosis as one of the candidate metrics. Our previous animal studies show that both kurtosis and energy are necessary to evaluate the hazard posed to hearing by a complex noise exposure. In this study, we focus on how to use the knowledge from animal model into humans. Methods are presented to solve the following questions: (1) How to calculate the kurtosis? (2) What is the relation between kurtosis and the hearing trauma? (3) How to extract a single number from the distribution of kurtosis that would best correlate with noise trauma in real industrial noise environments? Human data study shows that the kurtosis metric may be a reasonable candidate for evaluating the risk of hearing trauma complex noise exposures.

9:15

2aNS3. Assessing acoustic reflexes for impulsive sound. Gregory A. Flamme, Stephen M. Tasko, Kristy K. Deiters (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., MS 5355, Kalamazoo, MI 49008, greg.flamme@wmich.edu), and William A. Ahroon (Auditory Protection and Performance Div., US Army Aeromedical Res. Lab., Fort Rucker, AL)

The acoustic reflex is an involuntary contraction of the middle ear muscles in response to a variety of sensory and behavioral conditions. Middle ear muscle contractions (MEMC) have been invoked in some damage-risk criteria for impulsive noises for over 40 years and one damage-risk criteria proposes that MEMC precede the impulse for a warned listener via response conditioning. However, empirical data describing the prevalence, magnitude, and time-course of reflexive MEMC elicited by impulsive stimuli as well as non-acoustic stimuli and behaviors are scant. Likewise, empirical support for anticipatory MEMC is limited and studies often fail to control for attention or concomitant muscle activity. The current study is a large-scale, multi-experiment project designed to address these limitations in

a laboratory and field environment. MEMC are detected using click train stimuli as probes. Reflexive MEMC are elicited using tones, recorded gunshots, and non-acoustic stimuli (e.g., controlled release of compressed nitrogen gas to the face). Anticipatory MEMC are assessed across varying levels of distraction, beginning with participant instructions to pay attention to the conditioning stimulus and culminating in the assessment of anticipatory MEMC during live-fire exercises with rifles.

9:35

2aNS4. Measurement of high level impulse noise for the use with different damage risk criteria. Karl Buck (ISL (retired), 17 rue de la Résistance, Bartenheim 68870, France, kagebuck@aol.com), Pascal Hamery, Sebastien De Mezzo, and Florian Koenigstein (APC, ISL, Saint-Louis, France)

On the battlefield, but also during training, a soldier is continuously exposed to various types of noise (impulse and continuous). This exposure is not only noise generated by his own weapon but also by weapons or vehicles of close by troops. The exposure levels are between 160 dB peak for small arms and 190 dB peak at the soldier's ear for some anti tank weapons, with A-durations from 0.3 ms (small caliber) to 4 ms for large caliber weapons (e.g., Howitzers). In order to protect the soldier to noise exposures which may induce hearing loss, damage risk criteria (DRC) are implemented, and proposed for the prediction of the potential risk due to a certain noise exposure. Depending on the type of criteria (Pressure-Time-History or A-weighted Energy based), the recording and evaluation of different physical signal parameters has to be done in accordance to the used DRC. The paper will present the problems which may arise when recording impulse (weapon) noise with very high peak pressure levels and discuss measurement techniques compatible with the used DRCs. The paper will also discuss problems which may arise during the use and development of portable noise dose meters for the use in the military environment.

9:55

2aNS5. Damage risk criteria for high-level impulse noise and validation data. Armand L. Dancer (52 chemin Blanc, Authume 39100, France, armand.dancer@orange.fr)

The existing DRCs for high-level impulse noise will be briefly described along with their relative merits. Classical DRCs (CHABA, Mil-STD 1474D Z-curves, Pfander, Smoorenburg...) overestimate the hazard of large weapons, do not assess the actual efficiency of the HPs, and are not compatible with the occupational DRCs. The AHAAH model is potentially very powerful. However, the model needs to "know" the exact pressure-time history of the impulse at the subject's ear and the human middle ear transfer function for high-level impulses. Unexpected artifacts of measurements send the model on wrong tracks! Last but not least the "parameters" of this model need to be "adjusted" to be in agreement with the experimental results obtained on a large number of soldiers (to be presented). The LAeq8 method with a limit of 85 dB allows a limitation of the hearing hazard comparable to that aimed at by the other DRCs. It allows the assessment of the hazard for all kinds of weapon noises (free field and/or reverberant conditions) and for combined exposures (impulse and continuous noise) either on protected or unprotected ears. Finally, the auditory hazard is evaluated along the same rules in the military and in occupational exposures (ISO 1999).

10:15–10:30 Break

10:30

2aNS6. $L_{IAeq100ms}$, An A-duration adjusted impulsive noise damage risk criterion. Richard L. McKinley (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, richard.mckinley.1@us.af.mil)

Impulsive noise damage risk criteria (DRCs) have been the subject of much debate nationally and internationally for more than 30 years. Several approaches have been used in proposed DRCs including: curves defining exposure based on peak level and A or B duration; auditory hazard units based on the analytical auditory model known as AHAAH; and L_{Aeq} metrics based on the equal energy concept. Each of the approaches has positive and negative attributes. One of the issues with L_{Aeq} metrics has been the over estimation of hazard with long duration impulses such as those coming from blasts, artillery, large mortars, or shoulder launched missiles. The presentation will describe and discuss the $L_{IAeq100ms}$ impulsive DRC which includes an adjustment based on the A-duration of the impulsive noise and a method of computing protected impulsive noise exposures using impulsive insertion loss data from hearing protectors measured with methods defined in ANSI S12.42.

10:50

2aNS7. Development of the auditory hazard assessment algorithm for humans model for accuracy and power in MIL-STD-1474E's hearing analysis. G. R. Price (Human Res. & Eng. Directorate, Army Res. Lab., PO Box 368, Charlestown, MD 21914, ahanalysis@comcast.net) and Joel T. Kalb (Human Res. & Eng. Directorate, Army Res. Lab., Aberdeen Proving Ground, MD)

MIL-STD-1474E uses the Auditory Hazard Assessment Algorithm for Humans (AHAAH) model to calculate hazard from intense impulses, its theoretic basis providing greatly increased power and accuracy. It is an electro-acoustic analog paralleling the ear's physiology, including the ear's critical non-linearities and it calculates hazard from basilar membrane displacements. Successfully developed and tested first with an animal model, a parallel version for the human ear was developed and validated with human data. Given a pressure history as input, AHAAH predicts hazard for the 95%ile susceptible ear. It also makes a movie that allows engineering insight into amelioration of hazard. Hearing protection is accommodated by using input from an acoustic manikin or by implementation of a mathematical protector model using REAT data to calculate input waveforms from free field data. The AHAAH model is also currently used by the Society of Automotive Engineers for calculation of airbag noise hazard, by the Israeli Defense Forces for impulse noise analysis and is being considered by ANSI's S3 Bioacoustics Committee Working Group 62 (Impulse Noise with Respect to Hearing Hazard) as a basis for an ANSI impulse noise standard.

11:10

2aNS8. A biomechanically based auditory standard against impulse noise. Philemon C. Chan (L-3 ATI, 10180 Barnes Canyon Rd., San Diego, CA 92121, Philemon.Chan@L-3com.com)

This paper addresses two critical issues in developing a damage risk criteria against impulse noise injury, namely, (1) difficulty in setting a threshold for large weapon noise involving hearing protecting devices (HPDs) and small arms noise involving no HPDs; and (2) having a standard procedure to account for the effects of HPDs. The rational way to resolve these issues is to develop biomechanical-based standard using a physics-based model. The Auditory Hazard Assessment Algorithm for the Human (AHAAH) is a biomechanical model that simulates the transmission of sound energy from free field through the ear canal and middle ear to the cochlea. Extensive research has been performed by subjecting the AHAAH to a rigorous verification and validation process. Findings show that the AHAAH middle ear is overly compressive and corrections were made to the annular ligaments parameters. The human data from the historical Albuquerque walk-up study with volunteers wearing HPDs were used to validate the model and develop the dose-response curve for the injury threshold. Calculations were then performed for the German rife noise tests with volunteers not wearing HPDs, and the predictions show excellent comparison with the injury outcomes, hence providing an independent validation of the revised model.

Contributed Paper

11:30

2aNS9. Nonlinearity in the auditory hazard assessment algorithm for humans. Paul D. Fedele and Joel T. Kalb (Army, DOD, 520 Mulberry Point Rd., Attn: RDRL-HRS-D, Aberdeen Proving Ground, MD 21005-5425, paul.d.fedele.civ@mail.mil)

The Auditory Hazard Assessment Algorithm for Humans (AHAAH) is a software application that evaluates hearing damage risk associated with impulsive noise (http://www.arl.army.mil/ahaah). AHAAH applies pressure response dynamics across the external, middle, and inner ear, to biomechanically model the ear's physical response to impulsive sound. Cumulative strain-induced fatigue in the cochlea's organ of Corti determines the risk of auditory hazard. AHAAH includes nonlinear behavior observed in stapes displacement and associated with the annular ligament in the middle ear. AHAAH's nonlinear behavior has been validated by Price (2007) based on human test results produced by Johnson (1966, 1993, and 1997). Presented analyses results show that the risk of hearing hazard cannot be predicted solely on the basis of waveform energy (A-weighted or not) or waveform peak pressure, because of the middle ear nonlinearity. The risk of hearing hazard does not necessarily behave monotonically with any summary waveform characterization. Although the AHAAH may seem complex, it analyzes response to the full time-dependence of the waveform to accurately analyze hearing damage risk through the nonlinear elements of the human middle ear.

TUESDAY MORNING, 3 NOVEMBER 2015 — DAYTONA, 8:35 A.M. TO 12:00 NOON

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Flow-Induced Vibration

Robert M. Koch, Chair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Chair's Introduction—8:35

Invited Papers

8:40

2aSA1. Recent Japanese research activities on flow induced vibration and noise. Shigehiko Kaneko (Dept. of Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, kaneko@mech.t.u-tokyo.ac.jp)

In this presentation, recent Japanese research activity on Flow Induced Vibration and Noise mainly done by Kaneko laboratory, Department of Mechanical Engineering, the University of Tokyo, will be presented. Topics cover vortex induced vibration related to Japanese fast breeder reactor Monju thermo-well in line oscillation, sloshing and sloshing damper in connection with the liquid separator designed for Floating Production, Storage and Offloading (FPSO) system, galloping and galloping damper used for cable stayed bridges, combustion oscillation of gas turbine combustor taking account of chemical reaction process, and pipeline acoustics leading to acoustic fatigue. In the end, 30 years of history of data base group activity in Japan Society of Mechanical Engineers (JSME) will be introduced.

9:10

2aSA2. Investigating coupled flow-structure-acoustic interactions of human vocal fold flow-induced vibration. Scott Thomson (Dept. of Mech. Eng., Brigham Young Univ.-Idaho, AUS 106c, Rexburg, ID 83460, thomsons@byui.edu)

Flow-induced vibration of the human vocal folds is a central component of sound production for voiced speech. During vocal fold oscillation, tightly coupled flow, structure, and acoustic dynamics form a system that is rich in multi-physics phenomena, such as large deformation and large strain of exceedingly flexible and multi-layered tissues, repeated collision between vocal folds, coupling between structural modal frequencies and acoustic resonances, and the presence of non-trivial flow features such as the Coanda effect, flow separation, and axis switching. One of the aims of voice production research is to better understand these physical phenomena. In this presentation, tools and techniques for studying vocal fold flow-structure interactions will be discussed. Synthetic vocal fold replicas that exhibit flow-induced oscillations comparable to those of the human vocal folds will be introduced. These replicas are fabricated using three-dimensional prototyping, molding, and casting techniques, in which the multi-tissue layer structure of the human vocal folds is simulated using multiple layers of silicone of differing material properties. Experimental techniques used to characterize replica dynamic responses will be presented. Computational models that include fully coupled fluid, solid, and acoustic domains to simulate vocal fold vibration will be introduced. Several applications of these models and approaches will be discussed.

9:40

2aSA3. Fluid structure interactions with multicell membrane wings. Manuel Arce, Raphael Perez, and Lawrence Ukeiley (Mech. and Aerosp. Eng., Univ. of Florida, MAE-A Rm. 312, PO BOX 116250, Gainesville, FL 32611, ukeiley@ufl.edu)

Flexible wing surfaces can be observed in many natural flyers and their use in small engineered flying vehicles has translated too many beneficial properties. These benefits have manifested themselves in the aerodynamic forces as well as flight stability which are an effect of how the flow and the membranes interact both statically and dynamically. In this work, time dependent particle image velocimetry and digital image correlation are used to study the fluid structure interaction for flow over a membrane wing. The wings examined here are multi-cell silicon rubber membrane wings which have a scalloped free trailing edge with different levels of pretension. The pretension effects the natural frequencies of the membranes and is shown to affect the extension magnitude and membrane motion frequency which are both also affected by aerodynamic loading. Examinations of the membranes through time based and frequency domain analysis motions shows they are highly correlated with the flow. The velocity measurements demonstrated the effects of membranes motion alter the characteristics of the flow over wing leading to changes in the overall aerodynamic properties such as the stall angle and the wake deficit.

10:10

2aSA4. Computational flow noise. Donald Cox, Daniel Perez, and Andrew Guarendi (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, donald.l.cox@navy.mil)

This work focuses on combining the capabilities of computational fluid dynamics with computational structural acoustics to enable the calculation of flow noise primarily for undersea vehicles. The work is limited to the non-coupled problem, where the flow calculations are made over a non-deforming boundary with the goal of calculating wall pressure fluctuations and using them as loads on a finite element structural acoustics model. The ultimate goal of this work is to develop the capability to calculate flow noise for three-dimensional undersea structures for which analytical approaches are not possible. Results will be presented that make use of wall pressure fluctuations calculated using Large Eddy Simulations (LES) and variants of Improved Delayed Detached Eddy Simulations (IDDES).

10:40–11:00 Break

11:00

2aSA5. Using flow-induced vibrations for structural health monitoring of high-speed naval ships. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

It has been demonstrated theoretically and experimentally that an estimate of the impulse (or structural) response between two receivers can be obtained from the long-time average of the cross-correlation of diffuse vibrations (or ambient noise) recorded at these two receivers in various environments and frequency ranges of interest: ultrasonics, underwater acoustics, seismology, and structural health monitoring. Indeed, those estimated impulse responses result from the cumulated contributions over time of random vibrations (e.g., as created by flow-induced vibrations) traveling along the test structure and being recorded by both. Hence, this technique provides a means for structural health monitoring using only the ambient structure-borne noise (e.g., generated by flow-induce vibrations) only, without the use of active sources. We will review work conducted using (1) high-frequency random vibration (100 Hz–5 kHz) data induced by turbulent boundary layer pressure fluctuations and measured on a hydrofoil and a plate at the Navy's William B. Morgan Large Cavitation Channel. (2) Low frequency random vibration data (1 Hz–50 Hz) collected on high-speed naval ships during at-sea operations were strong wave impact loading took place. [Work sponsored by ONR.]

Contributed Papers

11:30

2aSA6. Wave dispersion in highly deformable, fluid-filled structures: Numerical and experimental study of the role of solid deformation and inertia. Patrick Kurzeja and Katia Bertoldi (John A. Paulson School of Eng. and Appl. Sci., Harvard Univ., 29 Oxford St., Pierce Hall, Rm. 410, Cambridge, MA 02138, patrick.kurzeja@rub.de)

The application and scientific interpretation of wave measurements in fluid-filled structures strongly depend on the frequency regime of interest. This includes, for example, absorption bands, inverse calculation of elastic moduli, and non-destructive crack localization. The wave properties significantly differ between the low-frequency regime (where viscous forces couple fluid and solid) and the high-frequency regime (where inertia forces allow for multiple decoupled wave modes with individual speeds and attenuation). Thus, knowledge of the separative transition frequency is crucial for a reliable prediction, but respective approximations like Biot's characteristic frequency still originate from stiff structures. Soft materials such as biological tissues or synthetic materials are neglected regarding their high deformability. Hence, this presentation demonstrates the change of wave properties from low to high frequencies in soft, fluid-filled structures and highlights the influence of solid deformability and inertia. In particular, it will present:an experimental design to control stiffness and density of a single porous structure by buckling mechanisms with negligible influence on permeability; microscale simulations to identify the underlying wave modes; and peculiarities occurring in soft fluid-filled structures such as significant dispersion of the P1-wave speed.

11:45

2aSA7. Experimental investigation of the acoustic damping of In-duct orifices with bias flow. Chenzhen Ji and Dan Zhao (Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Singapore 639798, Singapore, cji1@e.ntu.edu.sg)

Geometry of orifice is investigated by the experiments to evaluate the acoustic damping capacity of orifice plates in a duct. Four kinds of plates with complex orifice shapes are fabricated by using modern 3D printing technology. To characterize acoustic damping performance of these plates, sound absorption coefficient is used as an index determined by using the classical two-microphone technique. It is found that the geometric shapes of the perforated orifices can affect their sound absorption performances, and the damping performances of different shaped orifices depend on the frequency range. The length of downstream duct is also proven to determine the damping performance of perforated plates. The shorter the downstream pipe length, the narrower frequency range corresponding to lower power absorption. Moreover, the bias flow is shown to play a critical role on the sound absorption capacity of orifice plate in the experiments. Sound absorption coefficient is found to increase first and then decreased with increased Mach number.

TUESDAY MORNING, 3 NOVEMBER 2015 GRAND BALLROOM 8, 8:30 A.M. TO 10:00 A.M.

Session 2aSCa

Speech Communication: Speech Production Potpourri (Poster Session)

Sarah H. Ferguson, Chair
Communication Sciences and Disorders, University of Utah, 390 South 1530 East, Room 1201, Salt Lake City, UT 84112

Authors will be at their posters from 8:30 a.m. to 10:00 a.m. To allow authors an opportunity to view other posters in their session, all posters will be on display from 8:00 a.m. to 12:00 noon.

Contributed Papers

2aSCa1. Utterance-initial voiced stops in American English: An ultrasound study. Suzy Ahn (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu)

In English, phonologically voiced consonants are often phonetically voiceless in utterance-initial position. Other than Westbury (1983), there is little articulatory evidence regarding utterance-initial voicing in American English. The current study uses ultrasound imaging and acoustic measures to examine how tongue position correlates with phonation in American English, comparing phonated voiced stops, unphonated voiced stops, and voiceless stops in utterance-initial position. Eight speakers of American English recorded voiced/voiceless stops at three places of articulation (labial, alveolar, and velar), in three different environments (utterance-initial, post-nasal, and post-fricative), and with two different following vowels (high/low). One adjustment for initiating or maintaining phonation during the closure is enlarging the supraglottal cavity volume primarily via tongue root advancement. In utterance-initial position, there was a clear distinction between voiced stops and voiceless stops in the tongue root for the alveolar and velar places of articulation. Even without acoustic phonation during closure, the tongue root is advanced for voiced stops in comparison to voiceless stops for supraglottal cavity enlargement. These results suggest that speakers have the same target for both phonated and unphonated stops in utterance-initial position (i.e., shorter VOT), but other articulatory adjustments are responsible for the presence or absence of phonation.

2aSCa2. Aerodynamic factors for place-dependent voice onset time differences. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu), Mohammad Mehdi Alemi (Mech. Eng., Virginia Tech, Blacksburg, VA), and David J. Zajac (Dental Ecology, Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Studies have shown that voice onset time (VOT) tend to increase as place of articulation moves further back in the oral cavity. Different aerodynamic factors have been postulated for place-dependent VOT differences; although no direct aerodynamic measures have been reported in this regard.

The objective of this study was to investigate aerodynamic factors which lead to variation of VOT according to place of articulation. The speech materials of the study were /pa, ta, ka/, each produced 30 times by an adult female (27 yrs) in the carrier phrase "say – again". SPL was targeted within a ±3 dB range. Intraoral air pressure (Po) was obtained using a buccal-sulcus approach. VOT, Po, and maximum Po declination rate (MPDR) were measured for each stop. Results showed that: (a) the further back the place of articulation, the longer the VOT; (b) Po was greatest for the velar stop, intermediate for the alveolar stop, and smallest for the bilabial stop; and (c) the MPDR index showed slower pressure drop for the velar stop compared with the other two stops. Results provide empirical evidence for the role of oral pressure differences, mass of articulators, and cross-section area of the constriction in place-dependent variations of VOT.

2aSCa3. Effects of following onsets on voice onset time in English. Jeff Mielke (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu) and Kuniko Nielsen (Linguist, Oakland Univ., Rochester, MI)

Voice Onset Time (VOT) in English voiceless stops has been shown to be sensitive to place of articulation (Fischer-Jorgensen 1954), to contextual factors such as the height, tenseness, and duration of the following vowel and the voicing of coda consonants (Klatt 1975, Port & Rotunno 1979), to prosodic factors like stress and pitch (Lisker & Abramson 1967), and also to F0 (McCrea & Morris 2005) and speaking rate (Kessinger & Blumstein 1997, Allen 2003). We report two additional factors involving following consonants. We analyzed 120 /p/- and /k/-initial words produced by 148 Canadian English speakers (n = 17742). VOTs of the initial stops were measured semi-automatically and all other segment durations were measured using forced alignment. The results of a mixed-effects regression support earlier findings that VOT is longer in /k/, directly related to following vowel duration, inversely related to speech rate, longer before tense vowels, and shorter before voiceless codas. Additionally, we find that VOT is shorter when the next syllable starts with a phonetically voiceless plosive (i.e., excluding flapped /t/), and that the most relevant measure of vowel duration includes the duration of postvocalic liquids, even those that are typically analyzed as onsets.

2aSCa4. Individual interaction between hearing and speaking due to aging. Mitsunori Mizumachi (Dept. of Elec. Eng. and Electronics, Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, 804-8550, Japan, mizumach@ecs.kyutech.ac.jp)

It is well known that a hearing loss is induced by aging in a high frequency range. It is easy to imagine that the aging also alters characteristics of voice, because you can roughly estimate the speaker's age. In general, those aging phenomena are discussed independently. In speech communication, however, the speech chain [Dense & Pinson, 1993] must dominate the interaction on the aging effects between hearing and speaking. Individual interaction between them is investigated using both his pure-tone audiometry test threshold and his recordings of read utterances. In this study, 21 Japanese elderly males, whose ages ranged from 62 to 85 years old, participated in pure tone audiometry and recording of Japanese sentence and word utterances. Concerning three elderlies with presbycusis, who are aware of hearing loss in daily lives, hearing abilities gradually decrease in proportion to frequency over 2 kHz, and spectral energies increase in the high frequency range over 4.5 kHz. In another case of high-frequency deafness, the spectral energy over 4.5 kHz increases significantly. On the other hand, elderly speakers with normal hearing do not cause energy lift of speech in high frequencies.

2aSCa5. Degree of articulatory constraint predicts locus equation slope for /p,t,s,ʃ/. Sara Perillo, Hye-Young Bang, and Meghan Clayards (Dept. of Linguist., McGill Univ., Montreal, QC, Canada, sara.perillo@mail.mcgill.ca)

The degree of articulatory constraints (DAC) model (Recasens, Pallarès, & Fontdevila, 1997) proposes that consonants involving the movement of the tongue dorsum are more resistant to coarticulation than those with a more fronted articulation. We assessed this claim using locus equation (LE) slopes as indicators of coarticulation. Participants were asked to produce V1(t).CV2 sequences as part of two-word phrases in a scripted dialog, where C is one of /p, t, s, ʃ/. LE were derived by measuring *F2* at V2 onset and midpoint. Since LE slopes approaching 1 indicate high levels of coarticulation, it was hypothesized that segments with the lowest DAC would have the steepest slopes (/p/>/t/>/s/>/ʃ/) and this is what we found, lending support to the DAC model. A secondary hypothesis assessed the effect of emphatically stressing C on the LE. Participants partook in a dialog involving a "mishearing" of either the target C (Prominent condition) or the preceding V1(t) (Control condition), and they repeated the two word sequence. We expected participants to emphasize the misheard segment and reduce coarticulation if the C was misheard (lower LE slope). Our findings indicate that only the LE slopes of sibilants /s/ and /ʃ/ were reduced under prominence, perhaps due to their high DAC values.

2aSCa6. Effect of practice type on acquisition and retention of speech motor skills. Stephen M. Tasko (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W Michigan, Kalamazoo, MI 49008-5355, stephen.tasko@wmich.edu)

There is a growing literature focused on how speakers acquire and retain speech motor skills. While motor learning experiments provide practical information for improving speech treatment and instructional programs, identifying the specific conditions under which speech skills are enhanced or diminished also offers a window into the underlying organization of the speech motor system. The current study examines how speech motor performance on a challenging speech task varies for different forms of speech practice. Subjects include 40 healthy adult speakers. The challenging speech task is a set of tongue twisters produced at specified speech rates markedly faster than habitual rate. Subjects are assigned to one of three speech practice conditions (imitating an auditory target, listening to an auditory target, or using a magnitude production task) or a control task. Speech motor performance on the challenging speech task is assessed prior to, immediately after, and one day following the speech practice condition. Speech motor performance measures include speech rate accuracy and articulatory accuracy. Improved performance immediately following practice suggests speech skill acquisition while continued improvements during follow up testing suggests speech skill retention. The effect of different practice conditions on speech motor skill acquisition and retention will be described.

2aSCa7. Experimental validation of a three-dimensional finite-amplitude nonlinear continuum model of phonation. Mehrdad Hosnieh Farahani and Zhaoyan Zhang (Head and Neck Surgery, UCLA, UCLA Surg - Head & Neck, BOX 951794, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, mhosnieh@ucla.edu)

Due to the complex nature of the phonation process, simplification assumptions (e.g., reduced flow model, small strain vocal fold deformation) are often made in phonation models. The validity of these assumption is largely unknown because the overall behavior of these phonation models often has not been validated against experiment. In this study, a three-dimensional finite-amplitude nonlinear continuum model of the vocal folds is developed and compared to results from experiments using a self-oscillating physical model of the vocal folds. The simulations are based on a nonlinear finite element analysis, whereby large displacement and material nonlinearity are taken into account. The vocal-fold model is coupled with a reduced-order flow solver based on Bernoulli equation. Preliminary results show that the model is able to qualitatively reproduce experimental observations regarding phonation threshold and typical vocal fold vibration patterns. [Work supported by NIH.]

2aSCa8. A dual task study of the effects of increased cognitive load on speech motor control. Katherine M. Dawson (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gradcenter.cuny.edu), Grace Bomide (Speech-Language-Hearing Sci., Lehman College, New York, NY), Mark Tiede (Haskins Labs., New Haven, CT), and DH Whalen (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., New York, NY)

Research has shown that behavioral task performance suffers when cognitive load is increased. One method for observing this phenomenon is the so-called dual task paradigm, which has been applied in previous research manipulating the motor, linguistic, and cognitive demands of speech tasks [Dromey & Benson, *JSLHR, 46*(5), 1234–1246 (2003)]. Trade-offs between (and within) domains necessary to maintain task performance probe sensitivity to increased load and the nature of variability in speech (i.e., does speech become more or less variable in increased load situations). In the current study, cognitive load on a speech motor task (mono- and di-syllable repetition) is manipulated using simple, competing memory, visual attention, and inhibition tasks, with concurrent recording of speech acoustics and kinematics. A preliminary analysis of acoustic data from five participants measured duration, amplitude and F0 of utterances during the single and dual task conditions. Results from the dual task conditions suggest a complex trade-off between the amplitude, duration and F0 measures, which differed systematically among the memory, attention, and inhibition tasks. In contrast, the baseline speech measures varied idiosyncratically among participants. Analysis of kinematic data should assist in clarifying how the interactions among these variables are affected by the different types of cognitive load.

2aSCa9. Model based comparison of vocal fold dynamics between children and adults. Michael Döllinger, Denis Dubrovskiy, Eva Beck (Dept. for Phoniatrics and Pediatric Audiol. at the ENT Dept., Univ. Hospital Erlangen, Bohlenplatz, 21, Erlangen, Bavaraia 91054, Germany, michael.doellinger@uk-erlangen.de), and Rita Patel (Dept. of Speech and Hearing Sci., College of Arts and Sci., Indiana Univ., Bloomington, IN)

In clinical practice, pediatric vocal fold vibration patterns are visualized by methods and standards derived from the adult population. Quantitative evaluations of vocal fold vibratory changes which are connected to growth and development of children are missing, although it is known that the pediatric larynx is not simply a smaller version of the adult one. The aim of this study was to optimize the oscillations of a biomechanical two-mass-model (2MM) for children and adults and to judge whether dynamic differences exist. High speed recordings (4000 fps) at sustained phonation (vowel /i/) were recorded and analyzed. After glottis segmentation, vocal fold trajectories for 11 children and 23 adults (9 men, 14 women) were investigated. Model parameters were achieved by numerical optimization of the 2MM towards the vocal fold trajectories. Differences in oscillating masses, tissue stiffness, and subglottal pressure were identified and quantified. Children showed increased vocal fold stiffness as well as increased subglottal pressure values. Differences between children vs. men were more distinctive than between children vs. women. In summary, the study gives quantitative evidence of differences between pediatric and adult laryngeal dynamics and confirms the applicability of the 2MM towards children. Next steps will include analyses of disordered pediatric voices.

2aSCa10. Acoustic correlates of velar flutter associated with nasal emission during /s/ in children with velopharyngeal dysfunction. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu), Mohammad Eshghi (Inst. of TeleCommun. Systems, Technische Universität Berlin, Berlin, Germany), and David J. Zajac (Dental Ecology, Craniofacial Ctr., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Velar flutter can accompany obligatory nasal air emission in children with cleft palate. It usually occurs as a result of air passing through a partially closed velopharyngeal port that creates turbulence and tissue vibration due to aerodynamic-elastic forces. Nasal air emission can also occur without flutter. In this case, the velopharyngeal port is relatively large and turbulence is generated at the anterior nasal valve without flutter. In this study we applied auto-correlation to discriminate velar flutter from non-flutter nasal air emission. Three children with nasal turbulence and velar flutter and three children with nasal turbulence without flutter were recorded using the oral and nasal microphones of the Nasometer during production of /si/. Nasal emission of the /s/ sound captured by the nasal microphone was isolated and the auto-correlation functions of the signals were graphed using MATLAB. Results showed that nasal emissions with velar flutter have auto-correlation functions with periodic/quasi-periodic patterns. However, the auto-correlation functions of the non-flutter nasal emissions showed noisy fluctuations without periodic oscillations. Findings revealed that auto-correlation function can be used clinically as an acoustic technique to detect tissue vibration accompanied with nasal air emission.

2aSCa11. Deriving long-distance coarticulation from local constraints. Edward Flemming (Linguist & Philosophy, MIT, 77 Massachusetts Ave., 32-D808, Cambridge, MA 02139, flemming@mit.edu)

Coarticulatory effects can extend over two or more syllables. For example, we find in a study of English nonce words of the form [$bV_1C_1əC_2V_2t$] that F2 of V_1 is shifted toward the F2 of V_2. One approach to such long-distance coarticulatory effects posits direct interactions between the segments involved. For example, coproduction models attribute coarticulatory variation to temporal overlap between segments, so coarticulatory effects of V_2 on V_1 imply that a V_2 gesture begins two syllables earlier, during V_1. An alternative account posits that long-distance coarticulation results from iterative local coarticulation. That is, V_1 can show coarticulatory effects of V_2 because each intervening segment can partially assimilate to the next, resulting in a chain of coarticulatory effects between the two vowels. Since the iterative coarticulation analysis posits that long-distance coarticulation is mediated by intervening segments, it predicts that (i) coarticulatory variation at V_1 due to V_2 should be predictable from variation at the following segment, with no independent effect of later segments, and (ii) if intervening segments resist local coarticulation they should also attenuate non-local coarticulation across them. Neither prediction follows if distant segments can interact directly with each other. Both predictions are confirmed.

2aSCa12. Assessing vowel centralization in dysarthria: A comparison of methods. Annalise Fletcher, Megan McAuliffe (Dept. of Commun. Disord., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, annalise.fletcher@pg.canterbury.ac.nz), Kaitlin Lansford (School of Commun. Sci. & Disord., Florida State Univ., Tallahassee, FL), and Julie Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., Phoenix, AZ)

Previous literature has consistently reported correlations between acoustic vowel centralization and perceptual measurements of dysarthria. However, the strength of these relationships is highly variable, and many of the techniques used to measure vowel centralization have not been directly compared. This study evaluates methods of assessing vowel centralization and listeners' perceptions of dysarthria—with the aim of strengthening the relationship between these variables. Sixty-one speakers of New Zealand English (NZE; 17 healthy older individuals and 44 speakers diagnosed with dysarthria) read a standard passage. Metrics of vowel centralization were calculated using first and second formants of the [ɐː], [iː] and [oː] NZE point vowels. The results demonstrate that both the use of a flexible formant extraction point, and changes to the frequency unit in which formants are measured, can strengthen the relationship between acoustic and perceptual measures. Furthermore, applying these formant values to different metrics of vowel centralization, and changing the instructions listeners are given to rate dysarthria, can also reduce levels of unexplained variation in the relationship. In combination, these changes accounted for 18–26% more variance between vowel centralization measurements and listener perceptions of dysarthria in both male and female speakers.

Session 2aSCb

Speech Communication: Analysis and Processing of Speech Signals (Poster Session)

Alexander L. Francis, Chair
Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Authors will be at their posters from 10:30 a.m. to 12:00 noon. To allow authors an opportunity to see other posters in their session, all posters will be on display from 8:00 a.m. to 12:00 noon.

Contributed Papers

2aSCb1. Analysis of distinctive feature matching with random error generation in a lexical access system. Xiang Kong, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, 50 Vassar St. Rm. 36-523, Cambridge, MA, sshuf@mit.edu)

A matcher for a distinctive feature-based lexical access system is tested using degraded feature inputs. The input speech comprises 16 conversation files from a map task in American English, spoken by 8 female speakers. A sequence of predicted features are produced from a generation algorithm, and the results are randomly degraded at levels from zero to full degradation, for various combinations of the features. Two series of experiments are conducted: the first progressively degrades only single features while leaving all others intact, while the other builds up the system using single, then multiple features. From these experiments, introducing errors into particular articulator-free features, such as vowel, consonant, or sonorant; or articulator-bound features, such as the aspirated feature, pharyngeal features, the nasal feature, the velar feature, or the lateral and rhotic features, do not strongly degrade matching performance. However, matcher performance is more sensitive for errors in the other articulator-free features, and for the articulator-bound features related to vowel place and consonant place, especially, the tongue blade features. For combinations of features, degrading consonantal features, vowel place features, or tongue blade features leads to faster decline in performance, suggesting that these features play more important roles in lexical access.

2aSCb2. Suitability of speaker normalization procedures for classifying vowels produced by speakers with dysarthria. Kaitlin L. Lansford (School of Commun. Sci. and Disord., Florida State Univ., 201 W. Bloxham, Tallahassee, FL 32306, klansford@fsu.edu) and Rene L. Utianski (Dept. of Neurology, Mayo Clinic-Scottsdale, Scottsdale, AZ)

Speaker normalization, a process whereby the perceptual system of a listener recalibrates to accommodate individual speakers, is proposed to account for the ease with which we understand speech produced by multiple speakers with different sized and shaped vocal tracts. A variety of vowel-, formant-, and speaker-intrinsic or extrinsic transforms have been proposed to model speaker normalization of vowels produced by multiple speakers (e.g., Mel, Bark, and Lobanov methods). Suitability of such normalization procedures has been examined extensively in non-disordered speaker populations. Unknown at this point, however, is the appropriateness of normalization procedures for transforming spectrally distorted vowels produced by speakers with dysarthria. Thus, we examined the suitability of two transforms, Bark and Lobanov, for normalizing vowels produced by a heterogeneous cohort of 45 speakers with dysarthria. Non-normalized (Hertz) and Bark transformed vowel tokens were classified via discriminant function analysis (DFA) with 55% and 56% accuracy, respectively. Classification accuracy of vowel tokens normalized using Lobanov's method was 65%. The results of the DFAs were compared to perceptual data, which revealed listeners identified vowel tokens with 71% accuracy. These results suggest vowel-extrinsic and formant- and speaker-intrinsic normalization methods (e.g., Lobanov) are better suited to model speaker normalization of dysarthric vowels.

2aSCb3. Combining gestures and vocalizations to imitate sounds. Hugo Scurto, Guillaume Lemaitre, Jules Françoise, Frédéric Voisin, Frédéric Bevilacqua, and Patrick Susini (IRCAM, 1 Pl. Stravinsky, Paris 75004, France, hugo.scurto@ircam.fr)

Communicating about sounds is a difficult task without a technical language, and naïve speakers often rely on different kinds of non-linguistic vocalizations and body gestures (Lemaitre *et al*. 2014). Previous work has independently studied how effectively people describe sounds with gestures or vocalizations (Caramiaux, 2014, Lemaitre and Rocchesso, 2014). However, speech communication studies suggest a more intimate link between the two processes (Kendon, 2004). Our study thus focused on the combination of manual gestures and non-speech vocalizations in the communication of sounds. We first collected a large database of vocal and gestural imitations of a variety of sounds (audio, video, and motion sensor data). Qualitative analysis of gestural strategies resulted in three hypotheses: (1) voice is more effective than gesture for communicating rhythmic information, (2) textural aspects are communicated with shaky gestures, and (3) concurrent streams of sound events can be split between gestures and voice. These hypotheses were validated in a second experiment in which 20 participants imitated 25 specifically synthesized sounds: rhythmic noise bursts, granular textures, and layered streams. Statistical analyses compared acoustics features of synthesized sounds, vocal features, and a set of novel gestural features based on a wavelet representation of the acceleration data.

2aSCb4. Direct measurement of the dynamic range for rectangular speech passbands, from threshold to rollover. James A. Bashford and Richard Warren (Psych., Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Measurement of passband intelligibility can be confounded by appreciable contributions from transition bands under filtering conditions conventionally considered steep. Eliminating appreciable contributions outside of speech passbands can require slopes of several thousand dB/octave [Warren *et al*., JASA. 115, 1292-1295]. By employing effectively rectangular passbands, it is possible to determine their intrinsic intelligibilities, and also their dynamic ranges as determined by their threshold amplitudes and their decrease in intelligibility at high levels ("rollover"). Uncontaminated measures of these limits were obtained in the present study using 1-octave passbands (Experiment 1) and 1/3-octave passbands (Experiment 2) using rectangular speech bands (4800 dB/octave slopes) that spanned the frequency range from 0.25 to 8.0 kHz. Results obtained for the bands presented singly and in pairs, at levels ranging from threshold to 80 dB SL, indicate that [1] the speech dynamic range substantially exceeds 30 dB across most of the spectrum, and that [2] intelligibility rollover occurs at relatively low levels, exceeding approximately 70 dB. The use of rectangular speech bands for clinical assessment will be discussed. [Research supported by NIH.]

2aSCb5. Maintaining speech intelligibility at 100 dB using arrays of subcritical width rectangular bands. Richard Warren and Peter Lenz (Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, rmwarren@uwm.edu)

The Speech Intelligibility Index employs 16 contiguous 1/3-octave bands that sample the importance of frequencies across the speech spectrum. The present study employed the same Center Frequencies (CFs) using "Everyday Speech" sentences, but reduced the original 1/3-octaves having 26% bandwidths to 4% effectively rectangular bands (4800 dB/octave slopes). The resulting array of 16 subcritical-width bands had an intelligibility of 96% when heard at 60 dB despite having less than 16% of the 1/3-octave bandwidths. But, increasing the amplitude to 100 dB produced a decrease in intelligibility ("rollover") to 86%. In a parallel experiment, when the sixteen bands had a bandwidth of 40 Hz for each of their CFs, the intelligibility was 95% at 60 dB and decreased to 91% at 100 dB. But when a "chimera" or hybrid was created with a width of 40 Hz for all CFs from 0.25 kHz to 1 kHz, and a width of 4% for CFs from 1 kHz (bandwidth of 40 Hz) to 8 kHz (bandwidth of 320 Hz), then intelligibility was 99% at 60 dB, and 97% at 100 dB. Hybrids of this type may be of use in hearing aid design. [Research supported by NIH.]

2aSCb6. Arrays of subcritical width rectangular speech bands with interpolated noise maintain intelligibility at high intensities. Peter Lenz and James A. Bashford (Psychology, Univ. of Wisconsin - Milwaukee, PO Box 413, Milwaukee, WI 53201, plenz@uwm.edu)

Speech intelligibility declines at high intensities for both normally hearing and hearing-impaired listeners. However, this rollover can be minimized by reducing speech in high frequency regions to an array of noncontiguous bands having vertical filter slopes (i.e., rectangular bands) and widths substantially narrower than a critical band. Normally hearing listeners were presented with "Predictability Low" sentences consisting of a 500-Hz lowpass pedestal band and an array of ten 4% bands spaced at approximately 1/3-octave (alternate ERBn) intervals from 1000 Hz to 8417 Hz. The pedestal band was fixed at 70 dB and the subcritical-band array varied from 55 to 100 dB in peak level. Intelligibility did not vary significantly for levels from 65 to 95 dB, ranging from 86 to 89%. Array intelligibility did significantly decrease to 82% when speech level was increased to 100 dB. However, intelligibility was restored to 88% when lower level rectangular noise bands (−30 dB relative spectrum level) were interpolated between speech bands. It is suggested that subcritical width filtering reduces rollover by limiting firing rate saturation to a subset of fibers within critical bands, and that interpolated noise further reduces saturation via lateral inhibition. Hearing aid applications will be discussed. [Research supported by NIH.]

2aSCb7. Effect of depression on syllabic rate of speech. Saurabh Sahu and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland College Park, 8125 48 Ave., Apt. 101, College Park, MD 20740, ssahu89@umd.edu)

In this paper, we are comparing different methods to measure syllable rate of speech. Basically our method counts the number of vowels and divides it by the duration of speech. We use the energy content in 640–2800 Hz and 2000–3000 Hz to eliminate nasals and glides. Energy content in 0–400 Hz as well as pitch information and helps eliminate the unvoiced fricatives. We compare our method with Jong *et al*. (Behavior research methods. 2009; 41 (2): 385–390.) who wrote a Pratt script and with another method that estimates the syllable rate from peak modulation rate of speech. We have seen that the latter measure tracks the changes in HAMD scores and therefore seems sensitive enough to measure changes in the degree of depression. We will determine if these other methods will show the same sensitivity

TUESDAY MORNING, 3 NOVEMBER 2015 — CITY TERRACE 7, 9:00 A.M. TO 11:15 A.M.

Session 2aSP

Signal Processing in Acoustics: Detection, Feature Recognition, and Communication

Geoffrey F. Edelmann, Chair
U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375

Contributed Papers

9:00

2aSP1. Correlation trends in Naval Surface Warfare Center Panama City Division's database of simulated and collected target scattering responses focused on automated target recognition. David E. Malphurs, Raymond Lim, Kwang Lee, and Gary S. Sammelmann (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, david.malphurs@navy.mil)

In recent years, NSWC PCD has assembled a database of sonar scattering responses encompassing a variety of objects including UXO, cylindrical shapes, and other clutter-type objects deployed on underwater sand and mud sediments and inspected over a large range of aspect angles and frequencies. Data available on these objects consist of a simulated component generated with 3D finite element calculations coupled to a fast Helmholtz-equation-based propagation scheme, a well-controlled experimental component collected in NSWC PCD's pond facilities, and a component of measurements in realistic underwater environments off Panama City, FL (TREX13 and BayEX14). The goal is to use the database to test schemes for automating reliable separation of these objects into desired classes. Here, we report trends observed in an on-going correlation analysis of the database projected onto the target aspect vs frequency plane to clarify the roles of the environment, the data collection process, and target characteristics in identifying suitable phenomena useful for classification. [Work supported by ONR and SERDP.]

9:15

2aSP2. Doppler discrimination of a constant velocity scatterer at depth in shallow water. Christopher Camara, David Anchieta, Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), and Praswish Mahrajan (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, danchieta@umassd.edu)

Doppler as a discriminant for a shallow water moving rigid scatterer from a single element receiver is considered here. A mono-static source-receiver configuration emits a single tone to ensonify a moving object. Inference regarding the depth and speed of the moving object are sought from the amplitude and Doppler of the direct arrival and the surface interacting arrival. Computation of the full posterior probability distribution of the returned amplitudes and frequencies given the received waveform and the prior distribution on target depth is made by Markov Chain Monte Carlo sampling. A Gibbs sampler is employed to construct the posterior joint density of all parameters. Conditional and marginal densities of the amplitudes are analytically tractable while those of the frequencies are made with an importance sampling approach. Confidence intervals are computed and employed to address depth discrimination.

9:30

2aSP3. High-frequency, vertically directional short-range underwater acoustic communications. Geoffrey F. Edelmann, Lloyd Emokpae, and Simon E. Freeman (U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, WA, DC 20375, edelmann@nrl.navy.mil)

The underwater acoustic channel is a challenging environment for achieving high data rate communications due to multipath, attenuation, noise, and propagation delay. Performing and maintaining adaptive channel equalization requires significant computational overhead, leading to costly and power-hungry devices. Here we describe a low-cost reconfigurable acoustic modem platform (RAMP) intended to facilitate a cable-less benthic hydrophone array made from inexpensive and replaceable nodes. The high data rate acoustic modem is modulated on a carrier frequency of 750 kHz via binary phase shift keying (BPSK). Each modem is spaced approximately 10 m from adjacent units. Due to the vertical directivity of the transducer the half-maximum envelope of the main lobe is approximately 3°, thereby mitigating multipath from the bottom. Data at rates of up to 125 kbps will be shown from at-sea experimental measurements made in Panama City Beach, Fl. [This work was supported by the Office of Naval Research.]

9:45

2aSP4. Prediction of localization error in generating a focused source. Min-Ho Song (Musicology, Univ. of Oslo, Institutt for musikkvitenskap ZEB-bygningen 2. etg Sem Sælands vei 2, Oslo 0371, Norway, minho.song@imv.uio.no), Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea), and Yang-Hann Kim (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

This paper proposes a method of predicting human localization error for a focused source. A focused source is a virtual source located in between of a loudspeaker array and a listener. However, generation of the focused source cannot avoid the artifact due to causality, listeners always perceive pre-echoes before the desired sound. Since the human hearing system is sensitive to a preceding waves, it can lead a listener to perceive a virtual source in undesired direction. Because the repeating pre-echoes are observed for ineligibly long interval (~100 ms), it is not clear to distinguish timbral distortions and echoes from the localization error due to the summing localization of the human auditory system. Therefore, a suppression condition was defined from the precedence effect to separate the localization error from timbral distortions. After applying the suppression condition, the energy vector model was used to quantify the localization error. Combining the suppression condition and the energy vector model, localization error in horizontal plane for each listening spot considering positions of focusing point, array shape, driving solutions, spatial sampling, and truncation can be predicted. The examples show that the prediction method clearly holds up with focused source observations reported from relevant literature.

10:00–10:15 Break

10:15

2aSP5. Using automatic speech recognition to identify dementia in early stages. Roozbeh Sadeghian (Electrical and Comput. Eng. Dept., State Univ. of New York at Binghamton, 4400 Vestal Parkway East, Binghamton, NY 13902, rsadegh1@binghamton.edu), David J. Schaffer (ORC Inst. for Intergenerational Studies, State Univ. of New York at Binghamton, Binghamton, NY), and Stephen A. Zahorian (Electrical and Comput. Eng. Dept., State Univ. of New York at Binghamton, Binghamton, NY)

Early non-invasive diagnosis of Alzheimer's disease (AD) and other forms of dementia is a challenging task. Early detection of the symptoms of the disorder could help families and medical professionals prepare for the difficulties ahead, as well as possibly provide a recruitment tool for clinical trials. One possible approach to a non-invasive diagnosis is based on analysis of speech patterns. Subjects are asked to describe a picture and their description (typically 1 to 3 minute speech sample) is recorded. For this study, a database of 70 people were recorded, 24 with a clinical diagnosis of probable or possible Alzheimer's disease. When these data were combined with 140 other recorded samples, a classifier built with manually transcribed versions of the speech was found to be quite accurate for determining whether or not a speech sample was obtained from an Alzheimer's patient. A classifier built using automatically determined prosodic features (pitch and energy contours) was also reasonably accurate, with several subsets of pitch and energy features especially effective for classification, as assessed by cross validation. The manually transcribed text has now been replaced by automatically transcribed text using automatic speech recognition (ASR) technology. The main objective of this paper is to report on the relative effectiveness of several ASR approaches, including public domain ones, for this task.

10:30

2aSP6. Context recognition on smartphones by categorizing extracted features from acoustic and other sensor data. Miho Tateishi (School of Sci. for Open and Environment Systems, Keio Univ. Graduate School of Sci. and Technol., Hiyoshi 3-14-1, Kohoku-ku, Yokohama 223-8522, Japan, m.tateishi-3303@keio.jp)

This work presents a method of context recognition on smartphones using several built-in sensors. This system is developed for the Android. The three sensors are a microphone, an accelerometer, and a light sensor, which are correlated with human senses or movement. Context recognition is a method of being aware of the user's context or contiguous environment. In existing systems, raw time series data are directly used for the recognition. Our system aims to define data categories, which are connected with contexts, by extracting features from time series data. These features are made up of processed signals from each sensor. In particular, acoustic data turns into several different features; volume, spectrum average, peak spectrum appearance ratio, and correlation of these signals over a defined period of time. With acceleration and light features, it can classify similar contexts in the same category. For example, riding the train appears in one discreet category. Another category exemplifies the situation in a library, PC room, or laboratory which can be described as the "quiet" work place. As a part of automatic context awareness, features are formed hierarchical structure in order to make this method efficient.

10:45

2aSP7. The rear end collision and the wheel flying off protection using the reinforcement learning with the adaptive sound caution. Kazuhide Okada (Tele-Commun. Dept., College of Micronesia, P.O. Box 614, Daini, Kolonia, Kolonia, Pohnpei FM96941, Micronesia, rainbow_vc@yahoo.co.jp)

Safety driving is to watch both the front and the side from the driver seat. In order to avoid a rear end collision for the other car and a wheel flying off its axle for the said car itself at the same time, Q-learning of reinforcement learning was used in this study. In this study, the image of the front whole window is always captured by camera. Here, a steering angle divided by eight categories toward the front is the action a_t of a wheel flying off protection and car speed divided by ten categories is that of the rear end collision protection. The status s_t is distance from the sideview mirror to the side ditch for the wheel flying and the length between the roof of the car which is running ahead and its rear bumper. In the training phase, so as to maximize the sum of the rewards taken from the environment as the vehicle's front window, the

system updates the recursion Quantity of a state action combination $Q(s_t,a_t) = Q(s_t,a_t) + \alpha\ (r_{t+1} + \gamma\ \max_{[a]} Q(s_{t+1},a) - Q(s_t,a_t))$ (t:time, α:learning rate, γ: discount) periodically until its $Q(s_t, a_t)$ value is converged individually. When rewards both for the rear end collision and wheel flying off become mature, the automatic operating routines are regarded to be completed. And when making use of this system, the code still watches the car distance from the ditch and the relative level of the front car and the adaptive caution sound which has three spectrum peaks along 0.5 through 6 kHz are rung, adjusting frequency, loudness, and duty corresponding to the severity if the car approaches the crisis.

11:00

2aSP8. Stable QAM development with less BER on convergence time-compression type Q-learning for mass audio signal transmission. Kazuhide Okada (Tele-Commun. Dept., College of Micronesia, P.O. Box 614, Daini, Kolonia, Kolonia, Pohnpei FM96941, Micronesia, rainbow_vc@yahoo.co.jp)

This paper presents method which protects the temporary hang-up on the communication line and sustains the demodulation of the clear audible signal at the receiver, when the mass sound data is sent from the transmitter. QAM is one of the digital modulation technology, mapping the modulating signal not only toward the phase but also toward the amplitude on Q-I constellation, derived from QPSK. This modulation can pack larger data in the fixed period than QPSK. But once the communication path is exposed by stuff jitter or random jitter, the quantization error on its coordinate occurs, which means coordinate axes often subtly rotate with returning to the original position. In order to minimize such quantization errors on demodulation, Q-learning as one of Reinforcement Learning was used in this study. In the design of the feedback system comprehending the agent and the environment, the angle of the reverse vibration of the upper each axis for the quick restoring to the normal quantization becomes an action a_t. And the reward r_t is the relative baud rate, while the status s_t is BER, as I/F between Agent and the environment. The Quantity of a state action combination $Q(s_t, a_t)$ was updated as the index which measures the value of actions, with the Quantity computation steps decreased by TTD (Truncated Temporal Difference) and Log-time overlooking at $Q(s_t, a_t)$ in the training process of the experiment. The degree of control on coordinate's axes vibrations triggered by injected jitter was evaluated by the visible decrease in BER.

TUESDAY MORNING, 3 NOVEMBER 2015 GRAND BALLROOM FOYER, 9:00 A.M. TO 5:00 P.M.

Exhibit and Exhibit Opening Reception

The instrument and equipment exhibit is located near the registration area in the Grand Ballroom Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

The Exhibit will open on Monday with an evening reception with lite snacks and a complimentary drink.

Exhibit hours are Monday, 2 November, 5:30 p.m. to 7:00 p.m., Tuesday, 3 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 November, 9:00 a.m. to 12:00 noon.

Coffee breaks on Tuesday and Wednesday mornings (9:45 a.m. to 10:30 a.m.) will be held in the exhibit area as well as an afternoon break on Tuesday (2:45 p.m. to 3:30 p.m.).

The following companies have registered to participate in the exhibit at the time of this publication:

Brüel & Kjær Sound & Vibration Measurement—www.bksv.com

Freudenberg Performance Materials—www.Freudenberg-pm.com

G.R.A.S Sound & Vibration—www.gras.us

PCB Piezotronics—www.pcb.com/

Sensidyne—www.sensidyne.com

Springer—www.Springer.com

Teledyne Reson—www.teledyne-reson.com

Session 2pAAa

Architectural Acoustics and Musical Acoustics: Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio I

Timothy W. Leishman, Chair
Physics and Astronomy, Brigham Young University, N247 ESC, Provo, UT 84602

Chair's Introduction—1:20

Invited Papers

1:25

2pAAa1. Directional characteristics of musical instruments, and interactions with performance spaces. Jürgen Meyer (Braunschweig, Germany) and Uwe J. Hansen (Indiana State Univ., 64 Heritage Dr, Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu)

The seminal work: "Acoustics and the,Performance of Music" by Jürgen Meyer, translated by Uwe Hansen, includes a summary of decades of groundbreaking measurements in the acoustics laboratory of the PtB (Physikalisch-technische Bundesanstalt—Physical and Technical Federal Institution—Germany's Bureau of Standards). Interactions with faculty and students of the School of Music in Detmold, Germany, as well as with numerous audio engineers and performers have contributed to an understanding of the significance of these data. Directional characteristics of a number of musical instruments will be reviewed and discussed, as well as their effects on seating arrangement in the orchestra and interactions with performance spaces.

1:45

2pAAa2. Sound radiation properties of musical instruments and their importance for performance spaces, room acoustics measurements or simulations, and three-dimensional audio applications. René E. Caussé, Markus Noisternig, and Olivier Wausfel (Ircam - UMR STMS CNRS - UPMC, 1 Pl. Igor Stravinsky, Paris 75004, France, Rene.Causse@ircam.fr)

The directionality of the radiated sound is very specific to each musical instrument. The underlying radiation mechanisms may, for instance, depend on the structure of the vibrating body (e.g., string and percussion instruments) or on the spatial distribution of the opening holes (e.g., bells and open finger holes for wind instruments). A good knowledge of the radiation pattern of instruments is essential for many applications, such as orchestration, room acoustics, microphone techniques for live sound and recording, and virtual acoustics. In the first part, we will review previous works on sound source radiation measurement and analysis, discuss the underlying acoustic principles, and try to identify common mechanisms of radiation in musical instruments. In the second part, we will illustrate various projects undertaken at IRCAM and dedicated to the measurement and modeling of the directivity of instruments, to the objective and perceptual characterization of room acoustics, and to the real-time synthesis of virtual source radiation for musical performances. For this latter, several approaches are discussed according to the underlying physical formalisms and associated electroacoustic setups (e.g., spherical loudspeaker arrays, wave field synthesis).

2:05

2pAAa3. Database of musical instruments directivity pattern. Noam Shabtai, Gottfried Behler, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen D-52074, Germany, gkb@akustik.rwth-aachen.de)

The directivity pattern of an acoustic source describes the manner in which the sound radiates from it in the spatial domain. It may be used in virtual reality applications to improve the sense of realism perceived by the user. This work presents a directivity pattern database of 41 historical and modern orchestral instruments. The generation of this database includes the recording session in an anechoic chamber using a surrounding spherical microphone array, followed by a preliminary stage of isolating steady parts from the raw signals. Then, calibration is applied by normalizing the signals with the electrical channel gains and with the microphone gains. The fundamental frequency and overtones are then detected and the energy at each harmonic is saved for each played tone. Source centralization is applied in order to align the acoustic center of the sound source to the physical center of the microphone array. Last, a directivity pattern is generated in the spherical harmonics domain for each third-octave band by averaging the directivity pattern at all the overtones with a frequency belongs to that band.

2:25

2pAAa4. Challenges of musical instrument reproduction including directivity. Franz Zotter and Matthias Frank (Inst. of Electron. Music and Acoust., Inffeldfasse 10/3, Graz 8010, Austria, zotter@iem.at)

Reproduction of music from a solo instrument or singers by a single loudspeakers can suffer from lack of presence and liveliness if the directivity is missing. In particular, natural solo music contains the effects of a time-varying directivity with particularities in the directivity index, shape, yielding different coloration of diffuse reverb, early reflection, and direct sound. Focusing on the root of technical realization, it would be desirable to capture and reproduce a recording of an instrument played within a spherically surrounding microphone array in an anechoic chamber. This contribution reviews fundamentals (the soap bubble model) and technical solutions for this particular recording and playback problem, utilizing surrounding spherical microphone arrays and compact spherical loudspeaker arrays, also an application in which a trombone with directivity had been transmitted live from Graz to Paris. However, these solutions need to be carefully used as they hide some essential challenges: Surrounding spherical arrays suffer from the acoustical centering problem and the comb-filtering artifact it creates, and compact spherical loudspeaker arrays for directivity synthesis are subject to a trade off between bandwidth and spatial resolution against temporal resolution. For some of these challenges, alternative approaches will be addressed.

2:45

2pAAa5. Development, evaluation, and validation of a high-resolution directivity measurement system for live musical instruments. K. J. Bodon and Timothy W. Leishman (Phys., Brigham Young Univ., Provo, UT 84602, joshuabodon@gmail.com)

A measurement system has been developed to assess high-resolution directivities of live musical instruments. It employs a fixed, semicircular microphone array, a musician/instrument rotation system, and repeated note playing to produce 5-degree angular resolutions in both the polar and azimuthal angles. Its 2,522 spherical measurement positions reveal feature-rich, frequency-dependent directivity patterns. To date, a total of 16 wind and string instruments have been measured with the system. They were recorded as musicians repeated chromatic scales over standard working ranges following 5-degree rotations in the azimuthal angle, until a full revolution was completed. Directivity patterns of the first five partials of each note have been calculated and plotted as individual directivity balloons. While the approach provides high-resolution directivity results with reasonable numbers of microphones and data acquisition channels, it also has disadvantages, including lengthy recording and processing times. Special techniques have been developed to reduce the effects of nonideal measurement circumstances, including playing variances, musician movement, etc. A series of validation tests were performed using loudspeakers to simulate musicians under varying but controlled conditions. This presentation will discuss the methods and results of the work and provide comparisons to lower-resolution measurements.

3:05–3:20 Break

3:20

2pAAa6. Non-impulsive signal deconvolution for computation of violin sound radiation patterns and applications in sound synthesis. Alfonso Perez Carrillo, Jordi Bonada (Music Technol. Group, Universitat Pompeu Fabra, Roc Boronat 138, Barcelona 08018, Spain, alfonso.perez@upf.edu), Vesa Valimaki (Aalto Univ., Helsinki, Finland), Andres Bucci (Music Technol. Group, Universitat Pompeu Fabra, Barcelona, Spain), and Jukka Patynen (Aalto Univ., Helsinki, Finland)

This work presents a method to compute violin body impulse responses (BIR) based on deconvolution of non-impulsive signals. This newly conceived approach is based on a frame-weighted deconvolution of excitation and response signals. The excitation, consisting of bowed glissandi, is measured with piezoelectric transducers built into the violin bridge and the response is measured as sound pressure with microphones. Based on this method, several research works have been carried out in the areas of acoustics and sound synthesis. First, by placing multiple microphones at different angles around the violin, we were able to compute a dense grid of 3D sound radiation patterns without restrictions in the frequency range. Second, the computed BIRs can be convolved with a source signal (captured with the same bridge-transducer and using the same violin), obtaining a highly realistic violin sound very similar to that of a microphone recording. The multiple impulse responses at different directions make has been used to enhance sound synthesis with spatialization effects. Finally, a bowing machine was built to perform repeatable glissandi and therefore be able to compute BIRs across different violins. The bowing machine has been used to compute cross-BIRs that map the pickup signal of electric violins to the radiated acoustic sound of acoustic violins, which allows to imitate the sound of any measured acoustic violin with an electric counterpart.

3:40

2pAAa7. Influence of the instrumentalist on the sound of the concert harp. Jean-Loïc Le Carrou (Sorbonne Universités, UPMC Univ Paris 06, UMR CNRS 7190, LAM-D'Alembert, 11 rue de Lourmel, LAM - D'Alembert / CNRS / UPMC, Paris 75015, France, jean-loic.le_carrou@upmc.fr), Delphine Chadefaux (Aix-Marseille Univ., Inst. of Movement Sci., UMR CNRS 7287, Marseille, France), Baptiste Chomette, Benoît Fabre (Sorbonne Universités, UPMC Univ Paris 06, UMR CNRS 7190, LAM-D'Alembert, Paris, France), Francois Gautier (Laboratoire d'Acoustique de l'Université du Maine, UMR CNRS 6613, Le Mans, France), and Quentin Leclère (Laboratoire Vibrations Acoustique, INSA Lyon, Villeurbanne, France)

The sound of musical instruments comes from a subtle mix between its mechanical behavior and its interaction with the instrumentalist. For the concert harp, the instrumentalist defines the initial conditions which determine the vibratory contents of the strings. This vibration is then radiated through the soundboard and the sound-box over a range of 7 octaves. Besides, this radiation may be affected by the instrumentalist's physical presence next to the instrument. The aim of the talk is to show, thanks to ten years of research on the physics of the concert harp, how the instrumentalist and the instrument act in the sound from the instrumentalist gesture to the radiated sound. For that, specific set-up and models are carried out in order to carefully analyze each important step of the sound production : plucking, strings coupling, dynamical behavior of the soundboard and radiated sound. These studies are performed when the instrument is isolated, without the harpist, or in playing situation in a musical context. The results show, for instance, that the instrumentalist gesture is a part of the spectral content of the sound, whereas the instrument's design has consequences on the directivity of each string's partial.

4:00

2pAAa8. A study of variance of spectral content and sound radiation in timpani player. Brett Leonard and Scott Shinbara (School of Music, Univ. of Nebraska at Omaha, 6001 Dodge St., SPAC 217, Omaha, NE 68130, bleonard@unomaha.edu)

Timpani, although limited in pitch material, have the ability to produce many subtle colors, exceeding most other membranophones. One of the most notable characteristics of timpani is the perceptual “bloom” of the sound as distance is increased with the drum. Timpanists spend many hours working on control of these sounds and the bloom through application of intricate mallet technique and striking location. Anecdotal evidence of these differences is passed down through generations of teachers and students, but very little objective data exists about the actual sound of the drums, and the variations that occur between players. This study endeavors to reveal the objective differences between players and techniques, particularly as it relates to the “bloom” of the sound as you move away from the drum. Control of this directivity and expansion of perceived sound at a distance may be the single most important factor in the quality of a timpanist’s sound. Measurements are taken at different distances and locations around the drum for more than 15 different subjects, revealing a complex and interesting spectral patter radiating from the drum.

4:20

2pAAa9. Electric guitar—From measurement arrays to recording studio microphones. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St, Ste. 3, Lowell, MA 01854, alex@fermata.biz), Jim Anderson, and Agnieszka Roginska (New York Univ., New York, NY)

A joint research effort by the audio recording programs at the University of Massachusetts Lowell and New York University has made use of a 32-microphone measurement array in the quantification and visualization of the spectral radiation of musical instruments. Work to date has focused on electric guitar and piano. The measured directivities of the guitar amplifiers offer rich insight for the recording engineer. Traditional microphone selection and placement strategies formed over decades, before such data existed, are found to have merit. The data also shed light on those potentially unattractive microphone locations to be avoided. The measurements, taken with high spatial resolution, reveal a process for microphone placement as much as providing a window into showing exactly where to place them. Measurements of the acoustic radiation from electric guitar amplifiers reveal a spatial complexity that many recording engineers anticipate, and add valuable further insight.

Contributed Paper

4:40

2pAAa10. High-resolution measurements of speech directivity. Jennifer K. Whiting, Timothy W. Leishman, and K. J. Bodon (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC, Provo, UT 84606, lundjenny@comcast.net)

Directivity patterns of loudspeakers are often included in room acoustics simulation packages, but those of live sources are less common, partly because of the scarcity of reliable high-resolution data. In recent years, researchers at Brigham Young University have explored high-resolution directivities of musical instruments. Their methods have now been adapted to directivity measurements of live speech. The approach uses a semicircular array of 37 microphones spaced with five-degree polar-angle increments. A subject sits on a computer-controlled rotating chair with his or her mouth aligned at the axis of rotation and circular center of the microphone array. He or she repeats a phonetically balanced passage at each of 72 five-degree azimuthal-angle increments. Transfer functions between a reference microphone signal from the rotating reference frame and every array microphone signal enable computations of high-resolution frequency-dependent directivity balloons. Associated coherence functions allow judgment of frequencies for which directivity data can be trusted. This presentation discusses the results of these measurements and compares them to previous measurements of speech and singing-voice directivities. Animations of directivity balloons over frequency show a more complete picture of speech directivity than has been previously published.

Session 2pAAb

Architectural Acoustics and Noise: Measuring Sound Fields in Healthcare Environments

Gary Madaras, Cochair
Making Hospitals Quiet, 4849 S. Austin Ave., Chicago, IL 60638

James S. Holtrop, Cochair
AcoustiControl LLC, 2464 Taylor Road Suite 214, Wildwood, MO 63040

Chair's Introduction—2:55

Invited Papers

3:00

2pAAb1. A multinational comparison of measurement methods and metrics in acoustic standards and guidelines for healthcare environments. Gary Madaras (ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com)

The field of architectural acoustics is in the initial stage of a paradigm shift when it comes to quantifying sound fields in healthcare facilities. A growing number of practitioners and researchers are questioning whether existing acoustic measurement methods and metrics relate well to patient perception of quietness or medical outcomes. A multinational overview will be provided of various acoustic standards and guidelines for healthcare facilities. Acoustic measurement methods and metrics from different countries are compared to identify commonalities and discrepancies. An update is provided on the progress of the World Health Organization's revision of their 1999 Community Noise Guidelines, particularly the hospital noise section. Information is compiled and presented in order to begin the process of possibly defining new acoustic measurement methods and metrics that relate more strongly to patient perception of quietness and medical outcomes.

3:20

2pAAb2. Measurement of loud noncontinuous exterior noise sources on patients. James S. Holtrop (AcoustiControl LLC, 2464 Taylor Rd. Ste. 214, Wildwood, MO 63040, jim@acousticontrol.com)

A method to measure non continuous exterior noise sources such as helicopters, waste removal trucks, emergency vehicles, and semi-trucks will be presented. The noise levels that can be generated by these sources can exceed 90 dBA, which can impact patients within the hospitals. As hospitals are 24 hour facilities these sources of noise can happen during both daytime and nighttime hours. Data will be presented on the various acoustical methodologies to quantify this type of noise to access the impact on patients in the hospital.

3:40

2pAAb3. Permanent sensor-based hospital noise discovery. John Bialk (W2288 County Rd. E, Neshkoro, WI 54960, john@quietyme.com)

John Bialk, CEO of Quietyme, will discuss how using smart sensors and advanced analytics to measure sound levels in hospitals has revealed what really happens in healthcare settings. By measuring decibel levels once per second in every patient room, hallway, and nurse's station, Quietyme is able to uncover the exact sources of patient noise disturbances and better understand solutions to reduce them. With nearly 100 million points of hospital sound data, John Bialk has a rare and purely objective perspective on hospital noise and will explain the difference between what the data has revealed and common misconceptions. In addition, Quietyme has helped with a variety of noise studies testing rooms and John will reveal findings that relate to hard surface vs. carpet, healthcare noise levels during construction and more.

4:00

2pAAb4. Experiences evaluating acoustics in occupied hospitals. Erik Miller-Klein and Matthew Roe (SSA Acoust., LLP, 222 Etruria St., Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

The acoustic performance of patient rooms for four fully operational Veterans Affairs hospitals were completed over the past year. The nurse and facility managers goal was to identify the causes and solutions to improve the "quiet at night" scores. The testing included measuring speech privacy between patients and staff, background sound, occurrence rate of noise impacts during nighttime hours, and reverberation time. The testing procedures and methods were completed in cooperation with the hospital staff to optimize accuracy of the results while maintaining patient privacy, safety, and comfort. This included short duration pink noise measurements from room to room, and from nurses stations to patient care areas; these measurements provided accurate speech privacy metrics, though required direct approval and coordination with the nursing staff. The occurrence rate and background noise level was evaluated with 12 hour continuously logging sound level meters in patient rooms on tripods near patients with the approval of patients and guidance of nursing staff.

4:20

2pAAb5. Measuring quiet time in neonatal intensive care units. Jonathan Weber, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, jonryanweber@gmail.com), Ashley Darcy Mahoney (Nell Hodgson Woodruff School of Nursing, Emory Univ., Atlanta, GA), Myra Rolfes, Heather Cooper (Neonatal Intensive Care Unit, Children's Healthcare of Atlanta, Atlanta, GA), and Brooke Cherven (Nursing Res. and Evidence Based Practice, Children's Healthcare of Atlanta, Atlanta, GA)

The soundscape of critical care wards such as Neonatal Intensive Care Units (NICUs) are of particular concern due to the extremely sensitive nature of the patient population. NICUs must be conducive to providing care that enables infants to adapt to the extrauterine world without undue environmental stressors. Although the American Academy of Pediatrics (AAP) and others have set recommended noise limits in the NICU, studies consistently show units exceeding these standards. More nuanced aspects of NICU noise, such as source type, spectral content, fluctuations, and speech intelligibility are also of concern. A long-term study is being conducted that aims to improve NICU soundscapes, including measuring the impact of a Quiet Time (QT) evidence-based practice change. The study is a unique collaboration between engineering, architecture, nursing, and medicine. Detailed acoustic measurements were taken over a 18-month period to assess the soundscape in pre-QT, short-, mid-, and long-term post QT implementation periods. The study methodologies and results will be discussed, including considerations for the complexities of measuring sound fields in NICUs. Results are being used to identify and evaluate soundscape interventions and therefore advance understanding of how to design, measure, and implement healthy NICU soundscapes.

4:40

2pAAb6. How to identify and control noise that could cost your hospital money. Joe Mundell (Sonicu, 19 W. Main St., Greenfield, IN 46140, jmundell@sonicu.com)

Noise levels in hospitals are problematic. Today, patient opinions on noise, because of HCAHPS, determine hospital reimbursement. It's critical for hospitals to reduce noise levels in their facilities or face reductions in reimbursement. Patient perceptions of noise in hospitals is difficult to accurately measure and determine root cause. Many hospitals are implementing programs to address noise, but very few are making these decisions based on systematic sound monitoring data within their facility. Sound monitoring equipment installed in hospitals can accurately measure and identify both sources and patterns of noise in hospital. Real-time and historical sound data can help inform changes needed. We compare historic sound data from our experience monitoring sound levels in NICU's and compare findings to improving sound levels throughout hospital. We discuss how visual alarming alone is insufficient to maintain long term improvements. We also discuss the use of measuring "sound events" to idenitfy ture source of noise disturbances. We argue that hospitals that measure "sound events" can systematically improve their HCAHPS scores. In conclusion, sound recording equipment within hospitals will provide actionable data. This data can be used to implement structural and procedural improvements allowing hospitals to maximize reimbursement and improve outcomes.

Contributed Paper

5:00

2pAAb7. Opportunity for the session's participants to share their insights and lessons learned. Gary Madaras (ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com) and Jim Holtrop (AcoustiControl, St. Louis, MO)

This session on measuring sound fields in healthcare environments will conclude with an open microphone period whereby participants can contribute their brief insights and lessons learned from their past experiences performing acoustical measurements in healthcare settings. Contributions should be limited to measurement methods, standards, and interpretation of findings.

Session 2pABa

Animal Bioacoustics: Bioacoustics Across Disciplines: Emitting Sound

Philip Caspers, Chair
Mechanical Engineering, Virginia Tech, 1110 Washington Street, SW, MC 0917, Blacksburg, VA 24061

Contributed Papers

1:00

2pABa1. How flying CF-FM echolocating bats adapt to acoustically jammed environments: Quantitative evaluation. Daiki Goto, Shizuko Hiryu, Kohta I. Kobayasi, and Hiroshi Riquimaroux (Doshisha Univ., 1-3 Miyakotani Tatara, Kyotanabe 610-0321, Japan, dmo1003@mail4.doshisha.ac.jp)

Echolocating bats face the acoustical interferences from sounds of other conspecifics; they can fly without colliding each other and avoid surrounding obstacles. The purpose of this study was to reveal how CF-FM bats extract their own echoes in acoustically jammed environments. The Japanese horseshoe bats (*Rhinolophus ferrumequinum nippon*) were flown with conspecifics in the flight chamber. As the number of flying bats increased from one to seven, the duration of constant frequency (CF) components decreased, whereas the terminal frequency modulated (TFM) components were extended in both time and frequency ranges. In order to quantitatively evaluate behavioral responses under jamming conditions, a flying bat was exposed by artificially synthesized CF-FM pulses. As a result, the bats also changed the CF and TFM components as observed in the group flight experiment. This shows that the bats modify the characteristics of pulses to adapt to acoustical jamming and not to adapt to spatial jamming owing to other flying bats. These results suggest that TFM component was more important than CF component in extracting their own echoes during flight in acoustically jammed conditions. We will examine echolocation behavior when we manipulate CF components in context of jamming avoidance during Doppler-shift compensation.

1:15

2pABa2. Three-dimensional sonar beam control of FM echolocating bats during natural foraging revealed by large scale microphone array system. Kazuya Motoi, Miwa Sumiya (Life and Medical Sci., Graduate School of Doshisha Univ., Tataramiyakodani 1-3, Kyotanabe, Kyoto 610-0321, Japan, dmp1019@mail4.doshisha.ac.jp), Dai Fukui (The Univ. of Tokyo Hokkaido Forests Graduate School of Agricultural and Life Sci., Furano, Hokkaido, Japan), Kohta I. Kobayashi, Emyo Fujioka, and Shizuko Hiryu (Life and Medical Sci., Graduate School of Doshisha Univ., Kyotanabe, Kyoto, Japan)

In this study, 3-D flight paths and directivity pattern of the sounds emitted by *Pipistrellus abramus* during natural foraging were measured by a large scale microphone array system. The results show that the bats approached prey with covering the direction of them within their sonar beam. The means of horizontal and vertical beam widths were 49 deg and 46 deg, respectively. Just before capturing prey, the bats decreased the terminal frequency (TF) of the pulse. Simultaneously, the beam widths were expanded to 64 deg (horizontal) and 57 deg (vertical). We assumed a circular piston model to estimate how much the beam width was changed by decreasing the frequency of emitted pulse. It was found that the observed expansion of the beam width was smaller than those of theoretical estimations. This suggests that the bats decrease the TF of pulse for compensating their beam width narrowed by taking a large bite for the prey. We also measured echolocation calls and flight behavior of *Myotis macrodactylus* during natural foraging. *M. macrodactylus* uses FM echolocation pulse which is similar to *P. abramus*, but they forage for prey above the water surface. We compare echolocation strategies between two FM bats with different foraging habitat.

1:30

2pABa3. Dynamic baffle shapes for sound emission inspired by horseshoe bats. Yanqing Fu (Biomedical Eng. and Mech., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, fyanqing@vt.edu), Philip Caspers, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bat noseleaves are intricate baffle structures that diffract the animals' ultrasonic biosonar pulses upon emission. Furthermore, horseshoe bats dynamically change the shapes of their noseleaves through muscular actuation. Motions have been previously described for two lancet parts, anterior leaf and lancet. In both cases, the observed motions resulted in changes to the opening angle of the noseleaf baffle. Here, experiments were carried out with simplified baffle shapes that mimic the dynamics seen in horseshoe bats. For the baffle walls to have an effect on the outgoing wavefields, the sound outlets have to be narrow at least in one direction so that their near-fields generate substantial sound pressure amplitudes on the surface of the baffle. The baffle geometry was found to play an important role in the generation of dynamic signatures in the emitted pulses. As the opening of the baffle was varied by small increments, concave baffle surfaces were found to result in much larger dynamic changes to the beampatterns than straight baffles surfaces. Hence, concave baffles were able to introduce large dynamic signatures into the pulses even for small changes in opening angles. This may match the situation in horseshoe bats where the concerned baffles are also concave.

1:45

2pABa4. Numerical modeling of acoustic propagation in Harbor porpoise's (*Phocoena phocoena*) head. Chong Wei (College of Ocean & Earth Sci., Xiamen Univ., Hawaii Inst. of Marine Biology, Lilipuna Rd., Kaneohe, Hawaii 96744, weichong3310@foxmail.com), Whitlow W. Au (Hawaii Inst. of Marine Biology, Kaneohe, HI), Darlene Ketten (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Zhongchang Song (College of Ocean & Earth Sci., Xiamen Univ., Xiamen, China), and Yu Zhang (Key Lab. of Underwater Acoust. Commun. and Marine Information Technol. of the Ministry of Education, Xiamen Univ., Xiamen, China)

Harbor porpoises (*Phocoena phocoena*) use narrow band echolocation signals for locating prey and spatial orientation. In this study, acoustic impedance values of tissues in the porpoise's head were calculated from the Hounsfield Units (HU). A two-dimensional finite element model was set up base on the computed tomography (CT) scan data to simulate the acoustic propagation through animal's head. The far field transmission beam pattern in the vertical plane and the waveforms of the receiving points around the forehead were compared with prior measurement results, the simulation results were qualitatively consistent with the measurement results. The role of the main structures in the head such as air sacs, melon and skull in the acoustic propagation was investigated. Additionally, the relative sound pressure level within the porpoise's sonar field across the transitional near and far field was obtained to compare with the spherical spreading loss.

2:00–2:15 Break

2:15

2pABa5. Seeing the world through a dynamic biomimetic sonar. Philip Caspers (Mech. Eng., Virginia Tech, 1110 Washington St., SW, MC 0917, Blacksburg, VA 24061, pcaspers@vt.edu), Jason Gaudette (Naval Undersea Warfare Ctr., Newport, RI), Yanqing Fu (Eng. Sci. and Mech., Virginia Tech, Blacksburg, VA), Bryan Todd, and Rolf Mueller (Mech. Eng., Virginia Tech, Blacksburg, VA)

The outer baffle surfaces surrounding the sonar pulse emission and reception apertures of the biosonar system of horseshoe bats (family Rhinolophidae) have been shown to dynamically deform while actively sensing the environment. It is hypothesized that this dynamic sensing strategy enables the animal, in part, to cope with dense unstructured sonar environments. In the present work, a biomimetic dynamic sonar system inspired by the biosonar system of horseshoe bats has been assembled and tested. The sonar head features dynamic deforming baffles for emission (mimicking the bats' noseleaf) and reception (pinnae). The dynamic baffles were actuated to change their geometries concurrently with the diffraction of the emitted ultrasonic pulses and returning echoes. The time-variant signatures induced by the dynamic baffle motions were systematically characterized in a controlled anechoic setting and the interaction between emission and reception dynamic signatures was investigated. The sonar was further tested in the context of natural environments with a specific focus on the interaction of dynamic ultrasonic pulse packets with natural targets. For both experimental approaches, a sonar with static baffle shape configuration was used as a reference to establish the impact of dynamic features.

2:30

2pABa6. Use of vibrational duetting mimics to trap and disrupt mating of the Asian citrus psyllid, a devastating pest in Florida groves. Richard Mankin (Ctr. for Medical Agricultural and Veterinary Entomology, USDA ARS, 1700 SW 23rd Dr., Gainesville, FL 32608, Richard.Mankin@ars.usda.gov) and Barukh Rohde (Elec. and Comput. Eng., Univ. of Florida, Gainesville, FL)

The Asian citrus psyllid (ACP) is the primary vector of a devastating disease of citrus, huanglongbing, and efficient surveillance of ACP at low population densities is essential for timely pest management programs in Florida. ACP males search for mates on tree branches by producing vibrational calls that elicit duetting replies from receptive females. The males then search for the location of the reply. We constructed a vibration trap by using a microcontroller with signal detection software, a contact microphone to detect ACP calls, and a piezoelectric buzzer to produce calls. The buzzer plays back a female reply when a male calls, which stimulates the male to search and find it. In this report, we discuss the construction and operation of the vibrational trapping system. In addition, we discuss methods that have been developed in laboratory studies to interfere with ACP courtship and mating. Our goal is to develop field-worthy systems that target ACP infestations and reduce their populations.

2:45

2pABa7. Observations on the mechanisms and phenomena underlying vocalizations of the gray seals. Lukasz Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., ul. Pawinskiego 5B, Warszawa 02-106, Poland, lnowak@ippt.pan.pl) and Krzysztof E. Skora (Hel Marine Station, Univ. of Gdansk, Hel, Poland)

Gray seals vocalize both underwater and above the water surface using variety of sounds. The analysis of the differences in acoustic parameters of the emitted sounds suggests that the underlying phenomena may involve several different, independent mechanisms, which have not been yet investigated and understood. The aim of the present study is to introduce some important conclusions regarding those mechanisms and phenomena, based on the results of the original, long-term experimental investigations, and observations carried out at the sealarium of the Hel Marine Station of the University of Gdansk. Several thousands of vocalizations emitted by the mature specimens and cubs were recorded above and beneath the water surface, and analyzed for their acoustic parameters using the dedicated, developed software. The observations also involved video recording combined with synchronous acquisition of underwater sounds, which allowed to link the acoustic phenomena with the specific behavior of the animals. Based on the obtained results, the equivalent mechanical models of the corresponding anatomical structures responsible for generation of various sounds are proposed. An own, original classification of the vocalizations of the gray seals, based on the assumed separation of the involved generation mechanisms, is introduced.

3:00

2pABa8. Variations of fin whale's 20 Hz calls in the Gulf of California. Andrea Bonilla-Garzón (Biología Marina, Universidad Autónoma de Baja California Sur, Km 5.5 Carretera al Sur, Mezquitito, La Paz, Baja California Sur 23080, Mexico, nabonillag@unal.edu.co), Eduardo Romero Vivas (Centro de Investigaciones Biológicas del Noroeste CIBNOR,S.C., La Paz, Baja California Sur, Mexico), and Jorge Urbán-Ramírez (Biología Marina, Universidad Autónoma de Baja California Sur, La Paz, Baja California Sur, Mexico)

The fin whale's song has been broadly described in different areas of the world, and it is well characterized, being the spectrogram the main representation used to extract descriptions and measures. Males produce the 20 Hz calls, which consist of a down-swept pulse series (18–30 Hz) with a fundamental frequency of 20 Hz. Each pulse has duration of approximately 1 s, with aggrupation patterns of singlet, doublets and triplets. A time analysis of calls recorded by a High-frequency Acoustic Recording Packages (HARPs) localized in Punta Pescadero and Bahía de los Ángeles, south and north of the Gulf of California, México in the 2004 through 2008, revealed variations of the pulse that are not easily discernable through the use of the spectrogram. Results of the preliminary analysis of 100 calls are presented. Regional differences in duration, structure, and shape found could indicate a geographical separation of population units of fin whales in the Gulf of California.

3:15

2pABa9. The metabolic costs of producing clicks and social sounds differ in bottlenose dolphins (*Tursiops truncatus*). Marla M. Holt, Dawn P. Noren (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd. East, Seattle, WA 98112, Marla.Holt@noaa.gov), Robin C. Dunkin, and Terrie M. Williams (Dept. of Ecology and Evolutionary Biology, Univ. of California Santa Cruz, Santa Cruz, CA)

Dolphins produce many types of sounds known to have distinct qualities and functionalities. Whistles, which function in social contexts, are much longer in duration and require close to twice the intranasal air pressure to produce relative to biosonar click production. Thus, it is predicted that whistle production would be energetically more costly but this prediction is complicated by the fact that clicks are generated at much higher signal intensities. We used flow-through respirometry methods to measure metabolic costs of social sound and click production in two bottlenose dolphins. For all signal types, metabolic rates were related to the energy content of the signals produced. When metabolic costs were compared for equal energy sound generation, clicks were produced at negligible costs relative to resting and at a fraction of the cost of social sound production. However, while the performed repetition rates during click production were similar to field measurements, those of social sounds were much higher compared to typical field values. Even when metabolic costs are adjusted for more realistic whistle repetition rates, results indicate that whistle generation is more energetically costly. These results have implications for predicting the biological consequences of vocal responses to noise under different behavioral contexts.

Session 2pABb

Animal Bioacoustics: Bioacoustics (Poster Session)

Benjamin N. Taft, Chair
Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405

Authors will be at their posters from 3:30 p.m. to 5:15 p.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 1:00 p.m. to 5:15 p.m.

Contributed Papers

2pABb1. Mice ultrasonic detection and localization in laboratory environment. Yegor Sinelnikov (Acoust., Stevens Inst. of Technol., 126 Liberty Ave., Port Jefferson, NY 11777, ysinelnikov@yahoo.com), Alexander Sutin, Hady Salloum, Nikolay Sedunov, Alexander Sedunov (Acoust., Stevens Inst. of Technol., Hoboken, NJ), and David Masters (Dept. of Homeland Security, Sci. and Technol. Directorate, WA, DC,)

The acoustic detection and localization of mice movement by monitoring their ultrasonic vocalization has been demonstrated in laboratory environment using ultrasonic system with three microphones that provides recording of ultrasound up to 120 kHz. The tests were approved by Stony Brook University Institutional Animal Care and Use Committee protocol. Signals were recorded in a set of discrete sequences over several hours. The locomotor activity was characterized by durations up to 3000 ms and wide spectral content, while the syllable vocalization constituted shorter 200 ms events, with a set of identifiable up and down frequency modulated tones between 3 kHz and 55 kHz. The Time Difference of Arrival (TDOA) to various microphones was calculated using cross correlation method and was applied for estimation for mice location. Mice are among the invasive species that have a potential of crossing borders of United States unnoticed in containers. This study demonstrates the feasibility of using acoustic methods for detection of potential rodent intrusions. [This work was sponsored by DHS S&T.]

2pABb2. Effect of noise on each song element in Bengalese finch: Change of acoustic features. Shintaro Shiba, Kazuo Okanoya, and Ryosuke O. Tachibana (Graduate School of Arts and Sci., The Univ. of Tokyo, 3-8-1, Komaba, Meguro-ku, Tokyo 153-8902, Japan, shiba.shintaro@gmail.com)

Certain acoustic features of vocalization involuntarily change in response to the presence of background noise (*e.g.,* the Lombard effect). These changes, observed in many species, are considered to be induced by audio-vocal interaction. Bengalese finch (*Lonchura striata* var.*domestica*) is suitable for this study because they need real-time auditory feedback of their own song. By investigating the effect of noise on each distinct element (*i.e.* notes) of their song, we can get more detailed knowledge about audio-vocal interaction. Here we demonstrate the changes of intensity and fundamental frequency (F0) of notes of Bengalese finches' songs under noises. Two band-pass noises (High/Low) in two levels (Loud/Soft) were used. The High/Low noises had spectral bands of 4.0–7.8/0.2–4.0 kHz each, and the Loud/Soft noises were set to 70/60 dBA, respectively. As a result, intensity in two High conditions and F0 in High Soft condition increased, while these features of some notes decreased in Low conditions. Also some individual lowered F0 of almost all the notes in all the conditions. The results suggest that the change of these features might depend on the relation between noise frequency and the original note characteristics, and also individual tendency. [Work supported by JSPS KAKENHI #26240019.]

2pABb3. A study of Foley Sound based on analysis and compare of the bird's chirping. Ahn Iksoo (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro. Dongjak-gu. Seoul. Korea, Seoul 156-743, South Korea, aisbestman@naver.com), Seonggeon Bae (Daelim Univ., Anyang, South Korea), and Myungjin Bae (TeleCommun. & Information, Soolsil Univ., Seoul, South Korea)

This research verifies use value of Foley sound of bird‘s chirping used in Radio drama as sound contents by comparing and analyzing it with Actual sound. Radio drama utilizes sound of bird‘s chirping to show season, time, and place in various ways. Currently, this sound is used as sound effect in media by recording Actual sound of bird‘s chirping on site with portable recorder. Before when no portable recording device existed, they had to make Foley sound. To have comparative analysis of Actual and Foley sound of bird‘s chirping, production method of Foley sound tools and their usage were studied. To conclude, as Foley sound tools used for sound of bird‘s chirping are very unique and interesting there is a high possibility of developing them into Sound contents for Performance and Exhibition.

2pABb4. Rats became positive or negative states when listening to specific vocalizations. Yumi Saito (The Graduate School of Art and Sci., The Univ. of Tokyo, 3-6-17-302, Onitaka, Ichikawa, Chiba 2720015, Japan, ren_doll2000@yahoo.co.jp), Hiroko Kagawa (Brain Sci. Inst., RIKEN, Komaba, Japan), Shoko Yuki, and Kazuo Okanoya (The Graduate School of Art and Sci., The Univ. of Tokyo, Komaba, Tokyo, Japan)

Emotional contagion is the process in which one became the same emotional states of the others via a behavioral signal, and many species use acoustic cues as this signal. For example, in human beings, emotional contagion occurs when they get emotional vocalizations of others, like laughter or crying. Likewise, rats emit 50 kHz or 22 kHz specific ultrasonic vocalizations (USVs) associated with positive or negative context. In order to measure whether these acoustic cues have the valence of attractiveness or aversiveness, we utilized the cognitive bias task in rats. Rats were trained to respond differently for two neutral stimuli. One resulted in a positive outcome while the other produced negative outcome. Then the intermediate stimulus between the two was used to test whether that was interpreted as being positive or negative. After the 50 kHz USVs stimulation, rat interpreted the neutral stimuli as being positive, while after the 22 kHz USV stimulation, the same stimuli were regarded as negative. Results suggest that USVs can bring about positive or negative emotional states of listeners, and our findings indicated that specific USVs can work as acoustic emotional signal in the conspecific. [Work supported by JSPS KAKENHI #23118003.]

2pABb5. Neural responses to songbird vocalizations in the nucleus taeniae of the amygdala in Bengalese finches. Tomoko Fujii (Dept. of Life Sci.,Graduate School of Arts and Sci., The Univ. of Tokyo, 3-8-1, Komaba, Meguro-ku, Tokyo 153-8902, Japan, 5152187362@mail.ecc.u-tokyo.ac.jp), Maki Ikebuchi (Cognition and Behavior Joint Res. Lab., RIKEN Brain Sci. Inst., Saitama, Japan), and Kazuo Okanoya (Dept. of Life Sci.,Graduate School of Arts and Sci., The Univ. of Tokyo, Tokyo, Japan)

Emotions are important psychological processes that induce adaptive behavior in response to communication signals. Many species of songbirds form complex social relationships and communicate with others through a large variety of vocal sounds. Thus, songbirds could be a good model to search for the neural basis of emotion especially in a context of communication. Anatomical studies have shown that the nucleus taeniae of the amygdala (TnA) in birds corresponds to the medial amygdala in mammals. While the amygdala is suggested to be involved in recognition of conspecific vocalizations in rats and bats, the function of the TnA still remains unclear. The present study aimed to explore auditory response properties of the songbird TnA by electrophysiology. We examined activity of the Bengalese finch TnA neurons during the presentations of conspecific and heterospecific vocalizations, as well as synthesized sound. We demonstrated for the first time that a population of TnA neurons exhibited selective auditory responses to songbird vocalizations. Our findings suggested involvement of the songbirds TnA for the recognition of communicative sounds. Further investigation into the TnA response properties should be fruitful in understanding the relationship between emotion and vocal signals in animals. [Work supported by JSPS KAKENHI Grant # 26240019.]

2pABb6. Relating click-evoked auditory brainstem response waveforms to hearing loss in the bottlenose dolphin (*Tursiops truncatus*). Krysta L. Gasser Rutledge (Program in Audiol. & Commun. Sci., Washington Univ. in St. Louis, Campus Box 8042, 660 South Euclid Ave., St. Louis, MO 63110, gasserk@wusm.wustl.edu), Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and James F. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA)

Hearing sensitivity in captive Atlantic bottlenose dolphins was assessed using a portable electrophysiologic data collection system, a transducer attached to the pan region of the mandible, and non-invasive recording electrodes. The auditory steady-state response (ASSR) was evoked using sinusoidal amplitude-modulated tones at half octave steps from 20–160 kHz and utilized to determine the upper frequency limit of hearing (i.e., the frequency at which threshold was $\leq$ 120 dB re 1 μPa). An auditory brainstem response (ABR) was then recorded to a moderate-amplitude click (peak-equivalent sound pressure level of 122 dB re 1 μPa) and examined to determine if relationships existed between the upper frequency limit of hearing and the waveform characteristics of the click-evoked ABR. The ASSR and click-evoked ABR were measured in 6 bottlenose dolphins with varying hearing sensitivity and frequency range of hearing. A significant relationship existed between click-evoked ABR wave amplitudes and the upper frequency limit of hearing. Test times for assessment using frequency-specific ASSR and click-evoked ABR were $\sim$45 minutes and 1 minute, respectively. With further definition of normative data, measurement of click-evoked ABRs could form the basis of an expedited electrophysiologic method for hearing screening in marine mammals.

2pABb7. Auditory sensitivity shift by attention in Mongolian gerbil. Hiroyuki Miyawaki, Ayako Nakayama, Shizuko Hiryu, Kohta I. Kobayasi, and Hiroshi Riquimaroux (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe-shi, Kyoto-fu 610-0394, Japan, dmp1015@mail4.doshisha.ac.jp)

Mongolian gerbil, *Meriones unguiculatus*, communicate with others by various sounds. About 80% of those sounds range over 20 kHz. A threshold of hearing in this range, however, was about 20 dB higher than their most sensitive frequency (1 to 16 kHz). We proposed a hypothesis that the auditory sensitivity heightened when gerbils communicated with others. In order to test the idea, a cochlear microphonics (CM) was recorded under various behavioral context as a measure of auditory sensitivity. Under paired condition, a subject was set with another gerbil. The condition simulated the subject gerbils in their group, and the subject was considered to pay attention to sounds of the company. The CM response under the situation was higher than under single situation. Then, we investigated if the CM increase was specific under the situation. Subject was trained to pay attention to a sound of conditioned stimulus by electric shock as a negative reinforcer. The sound worked as an alarm sound for gerbils. The CM response increased while the gerbil was paying attention to the sound. Those results suggest that the sensitivity of auditory periphery is raised by attention in various behavioral contexts.

2pABb8. Vocal plasticity after laryngeal nerve lesion in rodent. Hiroki Iwabayashi, Shizuko Hiryu, Kohta I. Kobayasi, and Hiroshi Riquimaroux (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe-shi, Kyoto-fu 610-0394, Japan, dmp1007@mail4.doshisha.ac.jp)

Several species of animals, human, bat, and songbird adaptively control and change their spectro-temporal structure of vocalizations depending on auditory feedback. On the other hand, most of the common laboratory rodent generally vocalize relatively simple sound, and have been regarded as having only limited or no vocal plasticity. In this experiment, we conducted unilateral mutilation of the inferior laryngeal nerve in Mongolian gerbil, *Meriones unguiculatus,* and recorded the effect of surgery and recovery of vocalization for about 2 months for evaluating their vocalization plasticity. Mongolian gerbil have several types of vocalization. A short ($<$50 ms) high frequency ($>$25 kHz) vocalization, called "greeting call," is the most commonly observed in their colony. After mutilation of the nerve, they stopped producing greeting call. From 18 days after surgery (18 DAS) recovery began; spectro-temporal structure of high frequency vocalization altered day by day, and on 66 DAS they were able to vocalized calls similar to greeting call. Currently, we are investigating role of audition on the recovery by combining the laryngeal neurotomy and auditory deprivation. Neomycin was administrated in order to deafen the gerbil. We will discuss if auditory feedback is capable to modify vocal output in the rodent.

2pABb9. Analysis of bearings of vocalizing marine mammals in relation to passive acoustic density estimation. Julia A. Vernon, Jennifer L. Miksis-Olds (Appl. Res. Lab, The Penn State Univ., The Graduate Program in Acoust., The Penn State Univ., State College, PA 16803, jav232@psu.edu), and Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, The Univ. of St. Andrews, St. Andrews, Fife, United Kingdom)

The use of passive acoustic monitoring in population density estimation of marine mammals is a current area of interest, providing an efficient and cost-effective alternative to visual surveys. One challenge that arises with this method is uncertainty in the distribution of individuals. With large arrays where instruments are placed randomly with respect to the animals, it is often assumed that animals are uniformly distributed with respect to the instruments; however with sparse arrays this assumption is likely violated and could lead to bias in the density estimates. Distribution can be better determined through consideration of the horizontal azimuths or bearings of vocalizing animals. This paper presents bearing estimates of fin whales around Wake Island in the Equatorial Pacific Ocean, using ambient recordings from the Comprehensive Nuclear-Test Ban Treaty Organization (CTBTO) hydrophones at this location. Bearings were calculated for calls detected automatically. Multiple automatic detectors were assessed for optimal performance. Spectrogram correlation was found to produce the best results and bearings were calculated on calls detected with this method. The bearings were calculated using time delay information from the cross-correlation of received signals. Seasonal variation in animal distribution is also discussed. [This work was supported by the Office of Naval Research.]

2pABb10. Pulsed sounds produced by amazon river dolphin (*Inia geoffrensis*) in the Brazilian Amazon: Comparison between two water turbidity conditions. Jéssica F. Melo, Thiago Amorim, and Artur Andriolo (Zoology, Universidade Federal de Juiz de Fora/ Federal University of Juiz de Fora R. José Lourenço Kelmer, s/n - Campus Universitário, Juiz de Fora, Minas Gerais 36036-900, Brazil, jessicafmelo@live.com)

Pulsed sounds produced by the amazon river dolphin compose their acoustic repertoire, and possibly have communicative function. We

analyzed the acoustic behavior of amazon river dolphin under two water turbidity conditions in the Brazilian Amazon. Data were collected during three days when animals exhibited foraging behavior. The sounds were classified according to spectrographic visual characteristics. The acoustic parameters were obtained for each category. The Wilcoxon test was applied to compare the acoustic parameters between black water (BW) and white water (WW). In a total of 525.47 minutes of recording in Juami-Japurá Ecological Conservation Unit, 70.6% was in black water and 29.4% in white water. We found seven types (A, B, C, D, E, F, and G) of pulsed sounds. Types A, C, D and F were found exclusively in black water, while B (BW 46.4%; WW 53.6%), C (BW 80.8%; WW 19.2%) and G (BW 50.7%; WW 49.3%) were present in both conditions. The type B showed significant differences ($p < 0.01$) in low frequency, center frequency and peak frequency. Types C and G showed no difference in the parameters between the waters. This result indicates that water turbidity plays a role on the acoustic behavior of amazon river dolphin.

TUESDAY AFTERNOON, 3 NOVEMBER 2015 RIVER TERRACE 2, 1:30 P.M. TO 4:55 P.M.

Session 2pAO

Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics: Passive-Acoustic Inversion Using Sources of Opportunity II

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, Fairfax, VA 22030

Karim G. Sabra, Cochair
Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Chair's Introduction—1:30

Invited Papers

1:35

2pAO1. Resolving small events within a dense urban array. Nima Riahi and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, nriahi@ucsd.edu)

We use data from a large 5200 element geophone array that blanketed 70 km^2 of the city of Long Beach (CA) to characterize very localized urban seismic and acoustic phenomena. Such small events are hard to detect and localize with conventional array processing techniques because they are only sensed by a tiny fraction of the array sensors. To circumvent this issue, we first identify significant entries in the large coherence matrix of the array (5200×5200 entries) and then use graph analysis to reveal spatially small and contiguous clusters of receivers with correlated signals. This procedure allows us to track local events over time and also characterize their frequency content. The analysis requires no prior medium information and can therefore be applied under conditions of relatively high scattering. We show how the technique exposes a helicopter traversing the array, several oil production facilities, and late night activity on a golf course.

2:00

2pAO2. Ships as sources of opportunity in acoustic source localization. Christopher M. Verlinden (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu)

Ships, which can be tracked using the Automatic Identification System (AIS), represent underutilized acoustic sources of opportunity (SOP) that can potentially be used to localize unknown sources, invert for environmental parameters, and extract information about the ocean environment such as the local time dependent Green's Function. An application of ships as SOP's used to localize targets is presented here. Rather than use replica fields for matched field processing (MFP) derived from acoustic models requiring detailed environmental input, data derived replicas from ships of opportunity can be used for assembling a library of replicas for MFP. The Automatic Identification System (AIS) is used to provide the coordinates for the replica library and a correlation based processing procedure is used to overcome the impediment that the replica library is constructed from sources with different spectra and will be used to locate another source with it own unique spectral structure. The method is simulated and demonstrated with field experiments. AIS information is retrieved from the Unites States Coast Guard Navigation Center (USCG NAVCEN) Nationwide AIS (NAIS) database.

2:25

2pAO3. Inversion of ship-noise scalar and vector fields to characterize sediments: A critical review of experimental results (2007–2014). Jean-Pierre Hermand (LISA - Environ. HydroAcoust. Lab, Université libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels, Brussels Capital 1050, Belgium, jhermand@ulb.ac.be)

During geophysical and ecosystem surveys in Mediterranean and Atlantic shallow waters, research vessels, fishing boats, and passenger ferries were used as acoustic sources of opportunity to characterize bottom sediments of different types including mud, fluid mud, and sand. The ships sailed a straight course at constant engine speed while a compact vertical array of pressure or pressure-gradient sensors, deployed from a small boat, sampled the generated noise field over a broad frequency range. Different acoustic observables were investigated that do not depend on the knowledge of ship noise characteristics to mitigate the impact of such uncertainty. For sole pressure measurement, extraction and parameterization of striation structures in range-frequency spectrograms at different receiver depths allow to determine effectively compression wave speed and thickness of a sediment layer as compared to a reference. For additional pressure gradient measurement, vertical impedances are estimated at different depths whose range and frequency dependence are highly sensitive to bottom properties and, in particular, density. Global optimization via genetic algorithm and sequential Bayesian estimation via particle filtering maximize the degree of similarity between predicted and measured impedance data. The paper will critically review the experimental results and compare them with ground truth data. [Work supported by ONR, FNRS-CNPq, WBI-CAPES, PREFACE project EC DG Env FP7.]

Contributed Paper

2:50

2pAO4. Estimation of low frequency sound attenuation in marine sediments. Rui Duan (Inst. of Acoust. Eng., Northwestern PolyTech. Univ., Xi'an, China), N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca), Kunde Yang, and Yuanliang Ma (Inst. of Acoust. Eng., Northwestern PolyTech. Univ., Xi'an, China)

This paper presents a method for estimating low frequency sound attenuation from information contained in normal modes of a broadband signal. Propagating modes are resolved using the time-warping technique applied to signals from light bulb sound sources deployed at relatively short ranges of 5 and 7 km in the Shallow Water '06 experiment. A sequential inversion approach is designed that uses specific features of the acoustic data that are highly sensitive to specific geoacoustic model parameters. The first feature is the modal group velocity which is inverted for sediment sound speed and sediment layer thickness. The second feature is the modal amplitude function which is inverted for water depth and receiver depths. The third feature is related to the modal amplitude spectrum and is inverted for source depth and sound attenuation. In each subsequent stage, estimates from the previous stage(s) are used as known values. The sequential inversion is stable and generates geoacoustic model parameter estimates that agree very well with results from other experiments carried out in the same region. Notably, the inversion obtains an estimated value of 0.08 dB/λ in the band 120–180 Hz for the de-watered marine sediment characteristic of the continental shelf at the site.

3:05–3:25 Break

Invited Paper

3:25

2pAO5. Inversion of normal mode functions and seabed properties using non-linear time sampling of low-frequency bowhead whale calls on a vertical array. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Cedric Arisdakessian (Ecole Centrale de Lyon, Brest, France), and Julien Bonnel (ENSTA Bretagne, Brest, France)

Previous research has demonstrated that individual modal components of signals propagating in an acoustic waveguide can be isolated from a single hydrophone using non-linear time sampling (Bonnel *et al*., 2010) and that mode shape functions can be directly extracted from vertical array data in a laboratory setting (Bonnel *et al*., 2011). Here, we apply and extend the technique to experimental data collected in 2010 from a 15-element vertical array, covering only 63% of the 53 m water column off the continental slope of the Beaufort Sea, during the fall bowhead whale migration. We demonstrate how up to five distinct mode shapes can be extracted from frequency-modulated bowhead whale calls, along with the vertical array tilt and tilt direction, without resorting to assumptions about mode orthogonality across the array aperture. The extracted mode shapes can then be used to estimate whale call depth. Filtered modes (and mode cutoff information) can also be used to invert for whale call ranges and ocean bottom properties. [Work supported by the North Pacific Research Board and permitted by Shell Exploration and Production Company.]

Contributed Paper

3:50

2pAO6. Environmental inversion using bowhead whale calls in the Chukchi Sea. Graham A. Warner, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), David Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper estimates environmental properties of a shallow-water site in the Chukchi Sea using bowhead whale calls recorded on an asynchronous cluster of ocean-bottom hydrophones. Frequency-modulated whale calls with energy in at least two (dispersive) modes were recorded on a cluster of seven hydrophones within a 5 km radius. The frequency-dependent mode arrival times for nine whale calls were used as data in a Bayesian focalization inversion that considered the whale locations and range-independent environmental properties (sound-speed profile, water depth, and seabed geoacoustic profile) as unknown. A trans-dimensional inversion over the number of points defining the sound-speed profile and subbottom layers allows the data to determine the most appropriate environmental model parameterization. The whale-call instantaneous frequency, relative recorder clock drifts, and residual-error standard deviation are also unknown parameters in the inversion which provides uncertainty estimates for all model parameters and parameterizations. The sound-speed profile shows poor resolution but the thickness and sound speed for the upper sediment layer are reasonably well resolved. Results are compared to an inversion of controlled-source (airgun) dispersion data collected nearby which showed higher environmental resolution.

Invited Papers

4:05

2pAO7. Passive acoustics as a tool for assessment of seagrass ecosystems. Paulo Felisberto and Sergio Jesus (Univ. of Algarve, Campus de Gambelas, Faro 8005-139, Portugal, sjesus@ualg.pt)

Seagrass meadows are important coastal ecosystems, being the most productive biomes on Earth. The amount of produced oxygen is a key parameter for assessing the seagrass ecosystem metabolism and productivity. Several experiments conducted in *Posidonia oceanica* seagrass meadows have demonstrated that using active acoustic methods one can monitor the oxygen production. The acoustic method is sensitive to bubbles production and provides estimates at ecosystem level, what is an advantage to other methods. Healthy seagrass ecosystems are populated by several noisy marine taxa such as fishes and crustaceans, therefore the ambient noise can be used to assess the ecosystem metabolism and productivity. During two periods of one week in October 2011 and May 2013, acoustic and environmental data was gathered at several locations over a Posidonia oceanica bed in the Bay of Revellata in Corsica (France). The diurnal variability pattern of ambient noise characteristics (frequency band, power, directivity, and number of noisy sources) shows an evident correlation with oxygen production as measured by independent methods (optodes). The ambient noise data were dominated by impulse like waveforms associated with marine taxa. This work discusses the challenges faced on using these waveforms to invert for oxygen production.

4:30

2pAO8. Field calibration—Distributed vehicles and sensors and the use of their acoustic communications transmissions for probing the medium. Paul Hursky (HLS Res. Inc, 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

Underwater vehicles and float of various kinds are often used to provide distributed sensing or monitoring of the medium in which they are deployed. Such cooperating platforms typically communicate with each other using high-frequency, broadband acoustic communications. Such transmissions may provide much more information than is carried in their acomms payload, if they are used as ranging signals for navigation and probing signals for imaging the medium they travel through. We have collected data to explore this concept during two experiments, Makai 2005 in Hawaii and Radar 2007 in Portugal. We have previously shown how channel estimates gleaned from high-frequency probe signals from a moving platform and received on a receiving sensor array could be used to improve localization at lower frequency. We expand upon this previous work to demonstrate how such channel estimates at the relatively sparse locations where probes were transmitted by a vehicle can be interpolated to flesh out the channel impulse response throughout the localization search space. We also discuss how such transmissions can be used in aid of clock synchronization and navigation.

Session 2pBA

Biomedical Acoustics and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications II

Guillaume Haiat, Cochair
Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de Technologie Superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Contributed Papers

1:00

2pBA1. Three-dimensional and real-time two-dimensional topological imaging using parallel computing. Etienne Bachmann, Xavier Jacob (Lab. PHASE, 118 Rte. de Narbonne, Toulouse 31062, France, etiennebachmann@hotmail.fr), Samuel Rodriguez (I2M, Bordeaux, France), and Vincent Gibiat (Lab. PHASE, Toulouse, France)

We present the Fast Topological IMaging that has shown promising results to quickly process a picture by sending an ultrasonic plane wave within an unknown medium. This imaging algorithm is close to adjoint-based inversion methods but relies on a fast calculation of the direct and adjoint fields formulated in the frequency domain. The radiation pattern of a transducer array is computed once and for all, and then the direct and adjoint fields are obtained as a simple multiplication with the emitted or received signals, in Fourier domain. The resulting image represents the variations of acoustic impedance, and therefore highlights interfaces or flaws. Real-time imaging and high definition visualization both imply an expensive computation cost, that led us to implement this method on GPU (Graphics Processing Unit). Thanks to a massively parallel architecture, GPUs have become for ten years a new way to implement high performance algorithms. We used interoperability between OpenGL and CUDA to enable a real-time visualization. Experimental results in 2D/3D obtained with scalar waves are presented. At this time, the method has been implemented for acoustic waves in fluids, with an initial homogeneous medium, but it can be extended to elastic media and more complex configurations.

1:15

2pBA2. Temperature dependence of shear wave speed in a viscoelastic wormlike micellar fluid. E. G. Sunethra K. Dayavansha, Cecille Labuda, and Joseph R. Gladden (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., P.O. Box 1848, University, MS 38677, sdayavan@go.olemiss.edu)

Wormlike micellar fluids are viscoelastic and can support shear waves. Phase transitions of the micellar aggregates are temperature dependent and can manifest as sharp changes in the shear wave speed as a function of temperature. In this work, the variation of shear speed with temperature of 200 mM CTAB/NaSal micellar fluid in a 5:3 ratio was studied. The dependence of shear wave speed on time between fluid synthesis and measurement was also investigated. Shear wave propagation through the fluid was observed as a time varying birefringence pattern by using a high speed camera and crossed polarizers and shear speed was calculated by edge tracking and wavelength measurement techniques. A gradual increase in shear wave speed was observed in the temperature range 20—40 °C. A phase transition was observed to occur between 6 and 7 °C. There was no evidence of variation of shear wave speed with time. The implications of the shear wave speed variation over a wide temperature range will be discussed.

Invited Papers

1:30

2pBA3. Optimized excitation for nonlinear wave propagation in complex media: From biomedical acoustic imaging to nondestructive testing of cultural heritage. Serge Dos Santos (Inserm U930 "Imaging and Brain", INSA Ctr. Val de Loire, 3, Rue de la Chocolaterie, Blois, Centre-Val de Loire F-41034, France, serge.dossantos@insa-cvl.fr), Martin Lints (Inst. of Cybernetics, Tallinn Univ. of Technol., Tallinn, Estonia), Nathalie Poirot (GREMAN, IUT de Blois, Blois, France), and Andrus Salupere (Inst. of Cybernetics, Tallinn Univ. of Technol., Tallinn, Estonia)

The interaction between an acoustic wave and a complex media has an increase interest for both nondestructive testing (NDT) applications and for biomedical ultrasound. Today, new optimized excitations are generated thanks to the analysis of symmetry properties of the system such as reciprocity, nonlinear time reversal (TR) and other pulse-inverted (PI) techniques. Generalized TR based NEWS (Nonlinear Elastic Wave Spectroscopy) methods and their associate symmetry skeleton will be taken as an example. Among these family of "pulse coded excitation," solitonic coding constitutes a new scheme in the sense that solitary waves are the best candidates for pulse propagation in nonlinear and layered media, such as tooth or skin. As another application of mixing properties in a wide frequency range, new broadband techniques are needed in the domain of the preservation of cultural heritage. The analysis of the composition of the stones is one of the key parameters in the study of aging historic buildings. The use of TR-NEWS based analysis combined with a FTIR-based system has shown a specific property of the tuffeau limestone where damaged sample contains calcite. In both domains, aging properties of complex medium are extracted thanks to the use of enhanced nonlinear signal processing tools.

1:50

2pBA4. Ultrasound molecular imaging of heterogeneous tumors to guide therapeutic ultrasound. Frederic Padilla (Labex DevWeCan, Lyon Univ., INSERM LabTAU Unit 1032, 151 Cours Albert Thomas, Lyon 69390, France, frederic.padilla@inserm.fr), Alexandre Helbert, Isabelle Tardy (Geneva Res. Ctr., Bracco Suisse SA, Plan-les-Ouates, Switzerland), Cyril Lafon, Jean-Yves Chapelon (Labex DevWeCan, Lyon Univ., INSERM LabTAU Unit 1032, Lyon, France), Mathew von Wronski, François Tranquart, and Jean-Marc Hyvelin (Geneva Res. Ctr., Bracco Suisse SA, Plan-les-Ouates, Switzerland)

Propagation through heterogeneous tissues may hamper the ability to correctly focus ultrasound, especially when targeting a tumor for drug delivery applications. To solve this issue, we propose to use ultrasound molecular imaging (USMI) for treatment planning of drug delivery triggered by focused ultrasound. In a model of orthotopic rat prostate tumor having heterogeneous B-mode appearance, we show that tumor extend can be precisely delineated with USMI targeting VEGFR-2 receptors. High intensity ultrasound bursts (Peak negative pressure −15 MPa) are then delivered within this delineated 3D volume to trigger the release of liposomal-encapsulated chemotherapy, by local initiation of inertial cavitation. Real time imaging show that cavitation is indeed restricted to the targeted area. In the animal group receiving both chemotherapy and cavitational ultrasound, a potentiation of the therapeutic effect of the drug is clearly observed. This study demonstrates experimentally that USMI is an effective imaging method to characterize heterogeneous tissues and to guide therapeutic ultrasound.

2:10

2pBA5. Structure-factor model for quantifying the ultrasound scattering from concentrated cell pellet biophantoms. Emilie Franceschini, Romain de Monchy (Laboratoire de Mécanique et d'Acoustique CNRS UPR 7051, Aix-Marseille Université, Centrale Marseille, LMA CNRS UPR 7051, 31 chemin Jospeh Aiguier, Marseille 13009, France, franceschini@lma.cnrs-mrs.fr), and Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York, NY)

Ultrasonic backscatter coefficient (BSC) measurements were performed on K562 cell pellet biophantoms with cell concentrations ranging from 0.6% to 30% using ultrasound in the 10–42 MHz frequency band. The concentrated biophantoms mimic densely packed cells with known concentrations and are thus simplified versions of real tumor. Three scattering models, the fluid-filled-sphere model (FFSM), the Gaussian model (GM), and the structure factor model (SFM), were compared for modeling the scattering of the biophantoms. The GM and FFSM assume sparse, independently, and randomly distributed scatterers and are thus suitable for modeling dilute media; however, the SFM does not contain these assumptions and therefore can model dense media accurately. First, a parameter-estimation procedure was developed to estimate scatterer size and acoustic impedance contrast (assuming that cell concentrations were known *a priori*) and thereby compare theoretical with measured BSCs for all studied concentrations. The SFM yielded scatterer-radius estimates of 6.4 μm, which were consistent with the cell radius measured by optical microscopy. Second, the ability of the three models to estimate the scatterer size and acoustic concentration was compared. These scatterer properties were predicted well by the SFM, whereas the GM and the FFSM underestimated cell size and overestimated acoustic concentration for the more-concentrated biophantoms.

2:30

2pBA6. Modified transfer function with a phase rotation parameter for ultrasound longitudinal waves in cancelous bone. Hirofumi Taki (Dept. of Electron. Eng., Graduate School of Eng., Tohoku Univ., 6-6-05, Aramaki-Aza-Aoba, Aoba-ku, Sendai, Miyagi 980-8579, Japan, taki@ecei.tohoku.ac.jp), Yoshiki Nagatani (Dept. of Electronics, Kobe City College of Technol., Kobe, Japan), Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan), Katsunori Mizuno (Inst. of Industrial Sci., The Univ. of Tokyo, Tokyo, Japan), Toru Sato (Graduate School of Informatics, Kyoto Univ., Kyoto, Japan), and Hiroshi Kanai (Dept. of Electron. Eng., Graduate School of Eng., Tohoku Univ., Sendai, Japan)

The through-transmission ultrasound signal in cancelous bone consists of two longitudinal waves, called the fast and slow waves. The conventional propagation model assumes that the wavefront of an ultrasound wave passing through a bone specimen is flat and that each wave arrives simultaneously at all the points on a transducer surface. To compensate for the waveform change caused by the effect of the uneven wavefront received at a flat transducer, we propose a new transfer function modified with a phase rotation parameter. We also propose a fast decomposition method that requires only 5 seconds using a laptop personal computer. The proposed decomposition method using the modified transfer function succeeded to separate the two waves and characterize each of them accurately, where the normalized residual intensity power was less than −20 dB in the experimental study when the bone specimen thickness was from 6 to 15 mm. In the simulation study, the normalized residual power was less than −20 dB when the specimen thickness was from 3 to 8 mm. These results show that the transfer function with a phase rotation parameter is valid and that the proposed decomposition method has a great potential to provide good indicators of osteoporosis in real-time.

2:50

2pBA7. Simulation technique and time-frequency analysis of the acoustic wave inside cancelous bone. Yoshiki Nagatani (Dept. of Electronics, Kobe City College of Technol., 8-3, Gakuen-higashi-machi, Nishi-ku, Kobe, Hyogo 651-2194, Japan, nagatani@ultrasonics.jp)

Since the cancelous bone inside long bones has porous structure, the acoustic wave propagating inside bone may bring us useful information for the diagnosis of osteoporosis or bone healing although it is difficult to be investigated. In this paper, simulation techniques for understanding the complex wave propagation will be presented. The three-dimensional finite-difference time-domain (FDTD) method is useful for the wave simulation inside the complicated network structure. Due to the alignment of trabeculae, the pulse wave propagating inside cancelous bone separates into two wave, called the fast wave and slow wave. Here, the effect of the elasticity on the two wave phenomenon will be shown as well as the effect of the viscoelasticity. In addition to the speed and amplitude of the two waves, the frequency of the waves also changes corresponding to the structure of the medium. However, the detailed properties of the two waves are hardly revealed stably because of the wave overlapping. Therefore, a stable technique for quantitative evaluation of the wave characteristics is important. In this paper, the procedure and examples of a robust method of time-frequency analysis, named multi-channel instantaneous frequency, will be presented. These techniques help us understand and analyze the detailed wave behavior.

3:10–3:25 Break

Contributed Paper

3:25

2pBA8. Quantitative ultrasound measurements in trabecular bone using the echographic response of a metallic pin: Application to spine surgery. Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr), Yoshiki Nagatani (Kobe City College of Technol., Kobe, Japan), Seraphin Guipieri, and Didier Geiger (Multiscale modeling and simulation Lab., CNRS, Creteil, France)

Bone quality is an important parameter in spine surgery, but it remains difficult to be assessed clinically. The aim of this work is to demonstrate the feasibility of a QUS method to retrieve bone mechanical properties using an echographic technique taking advantage of the presence of a metallic pin inserted in bone tissue. To do so, an experimental validation is performed and acoustical modeling is used in order to assess the influence of experimental errors. A metallic pin was inserted in bone tissue perpendicularly to the transducer axis. The echographic response of the bone sample was determined and the echoes of the pin inserted in bone tissue and in water were compared to determine speed of sound (SOS) and broadband ultrasonic attenuation (BUA), which was compared to bone volume fraction (BV/TV). A 2-D finite element model was developed to assess the effect of positioning errors. Moreover, the coupling of finite difference time domain simulation with high resolution imaging technique was used to understand the interaction between an ultrasonic wave and the bone microstructure. A significant correlation between SOS and BV/TV was found ($R^2 = 0.6$). The numerical results show the relative robustness of the measurement method, which could be useful to estimate bone quality intraoperatively.

Invited Papers

3:40

2pBA9. A combined modeling and experimental investigation of ultrasonic attenuation mechanisms in cancelous bone-mimicking aluminum foams. Lawrence H. Le (Radiology and Diagnostic Imaging, Univ. of AB, Edmonton, AB T6G 2B7, Canada, lawrence.le@ualberta.ca), Behzad Vafaeian (Civil and Environ. Eng., Univ. of AB, Edmonton, AB, Canada), Kim-Cuong T. Nguyen (School of Dentistry, Univ. of AB, Edmonton, AB, Canada), and Samer Adeeb (Civil and Environ. Eng., Univ. of AB, Edmonton, AB, Canada)

Bone is composed of cancelous and cortical bones. Cancelous bone, which is more complicated than cortical bone in terms of structure, is inhomogeneous, anisotropic, and porous. The trabeculae form interconnected network with viscous marrow filling the pores. The presence of trabeculae makes cancelous bone a highly scattering medium. Trabecular bone spacing is considered an important parameter to detect change in bone tissue microstructures. However, due to high porosity and heterogeneity of human cancelous bones, the underlying mechanisms of interactions between ultrasound and cancelous bones are still not fully understood. The water-saturated aluminum foams were previously studied for the suitability of cancelous bone-mimicking phantoms. The ligament thickness and pore size of the foam samples were very similar to those of human cancelous bones. Recently, we performed the micro-scale *elastic* modeling of broadband ultrasound traveling through the water-saturated aluminum foams using the standard Galerkin finite element method. The simulated results were compared with the experimental measurements using the derived broadband ultrasound attenuation coefficients. The results strongly suggested that wave scattering and mode conversion were the dominant attenuation mechanisms of ultrasound propagating in aluminum foams. The study further demonstrated the capability of the finite element method to effectively simulate the signatures of the ultrasound signals propagating in fluid-filled weakly absorptive porous structures.

4:00

2pBA10. Ultrasonic guided waves in bone system with degradation. Dhawal Thakare (Mech. Eng., Indian Inst. of Technol. Madras, 410 MDS, Chennai, TN 600036, India), Pierre Belanger (Mech. Eng., Ecole de Technologie Superieure, Montreal, QC, Canada), and Prabhu Rajagopal (Mech. Eng., Indian Inst. of Technol. Madras, Chennai, TN, India, prajagopal@iitm.ac.in)

This paper investigates the feasibility of using ultrasonic guided waves for assessing the cortical bone and hence detect conditions such as osteoporosis. Guided wave propagation in bone systems modeled as multi-layered tubular structures consisting of anisotropic bone filled with viscous marrow and surrounded by tissue is studied using the Semi Analytical Finite Element (SAFE) method. Effects of changes to cortical bone thickness and mechanical properties are investigated. An attempt is also made to consider bone anisotropy in the models.The results, validated by experiments with bone phantoms, show that material and geometric condition strongly impacts the velocity of guided waves supported in the bone system. Identification of optimal guided wave modes for practical assessment is also discussed.

4:20

2pBA11. Axial transmission assessment of trabecular bone at distal radius using guided wave. Daniel Pereira, Alexandre Abid, Lucien Diotalevi (Dept. of Mech. Eng., Ecole de Technologie Supérieure, 6347 Christopher Colomb, Montreal, QC H2S2G5, Canada, pereira.ufrgs@gmail.com), Julio Fernandes (Ctr. de recherche l'Hôpital du Sacré-Coeur de Montréal, Montreal, QC, Canada), and Pierre Belanger (Dept. of Mech. Eng., Ecole de Technologie Supérieure, Montreal, QC, Canada)

The diagnosis of osteoporosis at skeletal sites composed mainly of trabecular bone, such as distal radius, can be considered clinically more relevant then cortical bone regions. Thus, the possibility to merge the potential of guided waves method to the clinical relevance of trabecular bone assessment is extremely motivating and has not yet been explored in details. Therefore, the objective of this paper is to investigate the feasibility of guided waves method to detect variation in the mechanical properties of trabecular bone at the distal radius using axial transmission. Due to the complexity of the distal region, three-dimensional finite elements simulations were performed using a bone model built from a computed tomography image of a human radius. The accuracy of the numerical model was experimentally validated using a bone phantom. The simulated excitation was transmitted to the radius using a longitudinal load applied to a small circular region at the extremity of distal radius. The identification of the guided waves was evaluated using 2DFFT transform. The results showed that the excitation imposed at the extremity of the distal radius creates guided waves that propagate along the long bone. Furthermore, the identified modes showed high sensitivity to the trabecular bone properties.

4:35

2pBA12. Guided wave velocity sensitivity to bone mechanical property evolution in cortical-marrow and cortical-trabecular phantoms. Alexandre Abid (Health Technology, Ecole de Technologie Supérieure (ETS), 3860 rue saint hubert appartement no. 1, Montreal, QC H2L 4A5, Canada, alexandreabid@hotmail.com), Dhawal Thakare (Indian Inst. of Technol. Madras, Chennai, India), Daniel Pereira (Mecanical, Ecole de Technologie Supérieure (ETS), Montreal, QC, Canada), Prabu Rajagopal (Indian Inst. of Technol. Madras, Chennai, India), Julio Fernandes (Res. Ctr. of Sacre Coeur Hospital, Montréal, QC, Canada), and Pierre Bélanger (Mecanical, Ecole de Technologie Supérieure (ETS), Montréal, QC, Canada)

Guided waves methods are sensitive to the mechanical properties of the material into which they are propagating. Guided waves methods are promising to detect osteoporosis; they are cost effective and do not expose the patient to radiation. Numerous studies have focused on the first arriving signal using a plate approximation. The aim of this study is to identify the best frequency/mode combinations for cortical-marrow and cortical-trabecular cylinder phantoms based on the sensitivity of the mode to mechanical property changes. Two different setups were used, axial transmission in a small diameter cortical-marrow phantom and circumferential propagation in a larger diameter cortical-trabecular phantom. Experiments for each method were carried out with in-plane and out-of-plane excitation, using frequencies from 50 to 200 kHz and were in good agreement with a 3D finite element model and dispersion curves extracted using semi-analytical finite element modeling. The modes' velocity was identified using either first zero crossing, short time Fourier transform or 2D Fourier transform. The evolution of the mechanical properties from a healthy bone to an osteoporotic bone was simulated using the finite element models. The results of this study will be used to further develop the technique using highly sensitive mode and frequency combinations.

4:50

2pBA13. An experimental study on the effect of temperature on the acoustic properties of cranial bone. Alec Hughes and Kullervo Hynynen (Dept. of Medical Biophys., Univ. of Toronto, 101 College St., Rm. 15-701, Toronto, ON M5G 1L7, Canada, ahughes@sri.utoronto.ca)

Transcranial focused ultrasound is increasingly being used as an alternative non-invasive treatment for various brain disorders, including essential tremor, Parkinson's disease, and neuropathic pain. These applications necessitate an understanding of the complex relationship between temperature and acoustic properties of cranial bone. In particular, the longitudinal speed of sound and attenuation coefficients will be investigated. In this study, *ex vivo* skull caps were heated to temperatures ranging from 20 to 50 °C, and ultrasound pulses were transmitted through the skull caps using a spherical transducer of 5 cm diameter and 10 cm focal length, at clinically relevant frequencies of 0.836 and 1.402 MHz. A thin Mylar film was placed at the focus, and a laser vibrometer was used to receive the ultrasound pulse transmitted through the skull. It was found that there was a measurable change in the phase and amplitude of the received signal, implying a change in both the speed of sound and attenuation of the bone at different temperatures. It was also found that these changes were completely reversible. These results imply that at sufficiently high cranial bone temperatures, the assumption of temperature-independent acoustic properties of bone may become invalid.

5:05

2pBA14. Assessing dental implant stability using quantitative ultrasound methods: Experimental approach and numerical validation. Romain Vayron and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr)

Dental implants are widely used for oral rehabilitation. However, there remain risks of failure that are difficult to anticipate. The objective of this study is to investigate the potentiality of quantitative ultrasound (QUS) to assess dental implant implant stability. To do so, the implant is initially completely inserted in the proximal part of a bovine humeral bone sample. The 10 MHz ultrasonic response of the implant is then measured and a quantitative indicator I is derived based on the *rf* signal obtained. Then, the implant is unscrewed by 2π radians and the measurement is realized again. The procedure is repeated seven times and the indicator is derived after each rotation of the implant. Analysis of variance (ANOVA) ($p < 10^{-5}$) tests revealed a significant effect of the amount of bone in contact with the implant on the distribution of I. In parallel, a finite element model is developed in order to model the ultrasonic propagation in the implant surrounded by bone tissue. The results show the feasibility of our QUS device to assess implant primary stability. A good agreement is obtained between experimental and numerical results. This study paves the way for the development of a new approach in oral implantology.

Session 2pEA

Engineering Acoustics: Analysis of Sound Sources, Receivers, and Attenuators

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

1:30

2pEA1. Experimental and computational multi-port characterization of a circular orifice plate. Stefan Sack and Mats Åbom (The Marcus Wallenberger Lab., The Royal Inst. of Technol., Teknikringen 8, Stockholm 100 44, Sweden, ssack@kth.se)

The generation and scattering behavior of fluid machines in duct systems is of great interest to minimize sound emission, for instance of air condition systems. Such systems may be described as linear, time-invariant multi-port networks containing passive elements, that scatter existing sound fields and active elements, that emit noise themselves. The aim of any multi-port analysis is to determine the direction-dependent transmission and reflection coefficients for the propagating wave modes and the sound generation. This parameters may be ascertained either numerically or experimentally. In a first step, external sound fields dominating the source are applied to determine the system scattering. In a second step, the source strength can be computed. Once these characteristic data are determined for all elements of interest, the sound scattering and emission of every considerable combination of multi-ports can be calculated easily. This paper shows an experimental study as well as a numerical approach to determine both, the passive and the active characterization of an orifice plate in a circular duct in presence of higher order acoustic modes. An enhanced measurement procedure is presented, and the results of this procedure are compared with data extracted from a full compressible DDES flow-computation.

1:45

2pEA2. Optimization of wave surface for planar loudspeaker array reproduction. Akio Ando (Faculty of Eng., Univ. of Toyama, 1-10-11 Kinuta Setagaya, Tokyo 157-8510, Japan, andio@a.memail.jp), Takuya Yamoto (Graduate School of Sci. and Eng., Univ. of Toyama, Toyama, Japan), and Daisuke Yamabayashi (Faculty of Eng., Univ. of Toyama, Toyama, Japan)

Over the past few decades, a considerable number of studies have been made on the sound field reproduction by loudspeaker array. It is well known that the reproduced sound field suffers from various errors caused by the finite and discrete loudspeaker array. The error by the finite array is a so-called truncation error, which is a kind of diffraction effect resulting from the absence of secondary sources outside the finite area. Such error sometimes brings the degradation of reproduced wave surface and frequency response. In this study, the weighting coefficient vector for loudspeaker input is introduced, meaning that the input signal fed into each loudspeaker is multiplied by an element of the vector. The input signal is calculated according to the Rayleigh I integral. The vector is solved such that the integration of square error between the ideal and synthesized wave surface over the target area is minimized under the constraint that all elements of the vector have non-negative values. Various optimization methods are compared in this problem. It is shown that the shape of wave surface is improved by the optimization and the improvement of wave surface also brings the improvement of frequency response.

2:00

2pEA3. A comparative study of acoustics and vibration analysis for wearing bearings. Bethany Snow, Chris Dunn, Lin Lin, and James Smith (Univ. of Southern Maine, 37 college Ave., Gorham, ME 04038, bethany.snow@maine.edu)

Monitoring of progressive wear and early diagnosis of damage are critical to a rotating machine to avoid dangerous, costly, time consuming failures. A common symptom of wear in rotating machines is characterized by an increase in noise and /or vibration levels. It is usually monitored by either accelerometers or acoustic emission sensors, both of which are required to be mounted on the machine. Acoustics measurement using microphones has its unique non-contact advantage of monitoring the sound that caused by the wearing. Some work has been done on investigating the power of the acoustic signals of wearing machines; however, the principal indicator of failure onset and the relation between the spectra of the vibration signal and the non-contact acoustic signal is unclear. In this project, both acoustic and vibration signals were measured to investigate a machine with bearings of different degrees of wear. Acoustic signal were recorded at various azimuth settings and compared to the vibration signal. Frequency analysis and signal processing techniques were applied to compare the information contents and sensitivities of those two measurements.

2:15

2pEA4. Nonlinear acoustic response of stand-off tubes used in acoustic pressure measurements—Part I: Experimental study. Anthony Rehl, John M. Quinlan, David Scarborough, and Ben Zinn (Aerosp. Eng., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30332, ajrehl@gatech.edu)

Rocket engine combustion instabilities, which lead to rapid engine failure through enhanced heat transfer rates and high-cycle fatigue, continue to be the most serious concern facing engine designers. Experimental testing and pressure measurements remains the best approach to determine the susceptibility of an engine design to acoustically coupled combustion instabilities. But, the harsh, high-temperature environment requires that pressure transducers be remotely mounted to the engine's main chamber using "sense-tubes," thereby creating an area-contraction at the connection point between the sense-tube and combustor. Preliminary measurements showed large discrepancies between sense-tube measured and engine acoustic pressure amplitudes. To elucidate these discrepancies, this experimental study measured the nonlinear response of the area-contraction/sense-tube geometry. Specifically, the sense-tube was attached to a two-microphone impedance tube, allowing the measurement of the acoustic impedance of the combined area-contraction and sense-tube; the acoustic pressure was also measured at the sense-tube termination, allowing the the direct measurement of the frequency response function. Measurements were performed over a range of frequencies, area-contraction ratios, acoustic velocity amplitudes, and sense-tube length-to-diameter ratios. These measurements reveal that the acoustic response of the sense-tube was highly nonlinear—even for low amplitude forcing.

2:30–2:45 Break

2:45

2pEA5. Nonlinear acoustic response of stand-off tubes used in acoustic pressure measurements—Part II: Analysis. John M. Quinlan, David Scarborough, Anthony Rehl, and Ben Zinn (Aerosp., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30332, jmquinlan@gmail.com)

Rocket engine combustion instabilities, which lead to rapid engine failure through enhanced heat transfer rates and high-cycle fatigue, continue to be the most serious concern facing engine designers. Experimental testing and pressure measurements remains the best approach to determine the susceptibility of an engine design to acoustically coupled combustion instabilities. But, the harsh, high-temperature environment requires remotely mounting dynamic pressure transducers to the engine's main chamber using "sense-tubes." Experiments reveal that the acoustic response of the sense-tube is highly nonlinear due to the area-contraction at the connection point between the engine and sense-tube. This nonlinearity leads to large discrepancies between the combustor's actual and the remotely measured acoustic amplitudes. Our aim was to develop an accurate, nonlinear sense-tube acoustic response model including steady flow effects. Therefore, the governing equation for the area-contraction pressure drop was approximated using a Fourier based technique to develop expressions for the steady and acoustic pressure drop across the area change. The acoustic pressure drop model was then incorporated into a response model for the tube. Measurements of the acoustic response of the area-contraction without mean flow agree well with predictions of the developed model.

3:00

2pEA6. Finite element method assisted development of an analytic model to describe the acoustic response of tee-junctions. Dan Fries and David E. Scarborough (Aerosp. Eng., Georgia Inst. of Technol., 635 Strong St., Atlanta, GA 30318, dfries@gatech.edu)

Tee-junctions are a central element of almost any duct system. The acoustics of such are altered by the transversal branch defining the tee. Resulting phenomena have been studied experimentally, numerically and analytically in the past. However, a simple practical tool for prediction and fundamental understanding is still missing. Applying the Finite Element Method, this work attempted to analyze the tee-junction's behavior with higher-order methods. To this end the commercial software COMSOL-Multiphysics was used. Thereafter, a low-order plane wave network model was developed and compared with simulation results. An error analysis of the power transmission coefficient computed by the model and the FEM simulation showed the limits of this low-order approximation. Thus, a guideline is provided for when the plane wave approximation delivers meaningful results. Moreover, it allowed for adjustments to the model increasing the model's accuracy, using a semi-empirical expression for a frequency dependent length correction of the side-branch. This had been predicted by other authors but never given explicitly. For side branches longer than two main-duct diameters and a thickness equal to or smaller than the main duct, the model gives good results up to 85% of cut-on of the first higher order mode. The proposed model can easily be implemented to estimate the characteristic quantities of a tee-junction.

3:15

2pEA7. A micromachined microphone based on a field effect transistor and an electret for low frequency acoustic detection. Kumjae Shin, Min Sung, and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technol., KIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk 790-784, South Korea, forhim13@postech.ac.kr)

Miniaturized microphone has sensitivity degradation in low-frequency region because most microphones use a capacitive-type transduction mechanism which has low-frequency cut-off. Therefore, detection of low-frequency sound is carried out using a microphone with large membrane and a large back chamber. To overcome this limitation, a micro-machined microphone based on a FET (Field effect transistor) and electret has been reported, and the feasibility of this system was demonstrated. The electric field due to the electret modulates the channel of the FET embedded in the membrane. The acoustic signal causes the FET mounted on the membrane to vibrate, which changes the separation between the channel of the FET and the electret. The resulting change in the electric field modulates the conductivity of the channel. The use of an integrated FET as a sensing mechanism in response to the electric field from the electret makes it possible to detect the displacement of the membrane directly. We describe a theoretical model for the low-frequency operation of this system and provide a comparison with experimentally measured results. A sensitivity analysis for the transduction mechanism is carried out, and the frequency response of the microphone is characterized using an acoustic measurement setup developed for low-frequency sound.

3:30

2pEA8. System and method for determining the level of a substance in a container based on measurement of resonance from an acoustic circuit that includes unfilled space within the container that changes size as substance is added or removed from the container. Robert H. Cameron (Icosahedron Corp., 714 Winter Dr, El Paso, TX 79902-2129, rcameron@elp.rr.com)

This abstract is for a poster session to describe a patent application made to the patent office in November 2012. The patent describes a system and method for determining the level of a substance in a container, based on measurement of resonance from an acoustic circuit that includes unfilled space within the container that changes size as substance is added or removed from the container. In particular, one application of this device is to measure the unfilled space in the fuel tanks of vehicles such as cars and trucks. For over 100 years, this measurement has been done by a simple float mechanism but, because of the development of tank design for vehicles that involve irregular shapes this method is increasingly less accurate. The proposed device will overcome these limitations and should provide a much more accurate reading of the unfilled space and therefore the amount of fuel in the tank since the total volume of the tank is known.

3:45

2pEA9. A stepped-plate radiator as a parametric array loudspeaker. Yonghwan Hwang and Wonkyu Moon (Postech, PIRO416, Postech, Hyo-ja dong, Nam gu, Po hang KS010, South Korea, serenius@postech.ac.kr)

A parametric array loudspeaker can generate a high directional audible sound beam in the air by the nonlinear acoustic interaction widely known as a "Parametric Array." The advantage of the parametric array phenomenon is that it creates a narrow beam using the small aperture of an acoustic radiator. It can be used in many applications, including high directivity communication systems. However, the sound pressure level generated by the parametric array is very low, which necessitates a high efficiency and intensity of sound. This paper describes the application of a stepped-plate radiator in a parametric array loudspeaker. The stepped-plate radiator has two resonance frequencies ($f_1 = 75$ kHz, $f_2 = 85$ kHz) in the design frequency region, and three different steps that compensate for the phase difference in the flexural vibrating plate. This allows the generation of high directivity audible sound in the wide flat radiation bandwidth of the audible frequencies range (100 Hz to 16 kHz, difference frequency waves with equalization). [Work supported by ADD (UD130007DD).]

TUESDAY AFTERNOON, 3 NOVEMBER 2015 DAYTONA, 1:30 P.M. TO 3:00 P.M.

Session 2pED

Education in Acoustics: Take 5's

Jack Dostal, Chair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

For a Take-Five session, no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session begins. If you have more than one demo, sign-up for two consecutive slots.

TUESDAY AFTERNOON, 3 NOVEMBER 2015 DAYTONA, 3:15 P.M. TO 4:45 P.M.

Session 2pID

Interdisciplinary: Guidance From the Experts: Applying for Grants and Fellowships

Caleb F. Sieck, Cochair
Applied Research Laboratories and Department of Electrical & Computer Engineering, The University of Texas at Austin, 4021 Steck Ave #115, Austin, TX 78759

Anna C. Diedesch, Cochair
Hearing & Speech Sciences, Vanderbilt Univ., Nashville, TN 37209

Michaela Warnecke, Cochair
Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St, Dept Psychological & Brain Sciences, Baltimore, MD 21218

Christopher Jasinski, Cochair
University of Notre Dame, 54162 Ironwood Road, South Bend, IN 46635

A panel of successful fellowship winners, selection committee members, and fellowship agency members organized by the Student Council.

This panel will consist of Laura Kloepper, post-doc at Brown University and recipient of an NSF Postdoctoral Fellowship, Alberto Rivera-Rentas, Research Training Officer for NIH/NIDCD, Andrew Oxenham, Professor at the University of Minnesota and NIH reviewer, and Jason Summers, Chief Scientist at Applied Research in Acoustics and recipient of grants from government institutions and private research foundations.

The panelists will provide a brief introduction about themselves and answer questions regarding grant and fellowship opportunities and application advice.

Session 2pNS

Noise: Damage Risk Criteria for Noise Exposure II

Richard L. McKinley, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Hilary L. Gallagher, Cochair
Air Force Research Lab., 2610 Seventh St. Bldg. 441, Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—1:00

Contributed Paper

1:05

2pNS1. Military Standard 1474E: Design criteria for noise limits vs. operational effectiveness. Bruce E. Amrein (Human Res. & Eng. Directorate, Army Res. Lab., ATTN: RDRL-HRS-D, 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005, amrein@gmail.com)

In April 2015, the U.S. Department of Defense (DoD) published a significant revision to Military Standard 1474 (MIL-STD-1474): Design Criteria—Noise Limits. Through the efforts of a DoD multi-service working group, every aspect of MIL-STD-1474 has been revised to improve readability; reduce conflicting guidance; and consolidate requirements common to steady-state and impulsive noise produced by weapons systems, and ground-, air- and water-borne platforms. Noise requirements in military environments differ significantly from typical industrial or occupational situations, and mission success requires offensive equipment and weapons to be more lethal and survivable than those used by the adversary. Producing material suitable for military operations requires unique design criteria often exceeding civilian national or international standards. MIL-STD-1474E provides the design tools and measurement techniques necessary to satisfy these unique and often contradictory requirements. For the first time in MIL-STD-1474E, a computer-based, electro-acoustic model of the human auditory system is used to evaluate hazard from impulsive noise events typical of weapon firing. This presentation describes the salient requirements necessary to produce and deploy military systems maximizing Warfighter effectiveness, while minimizing hearing damage caused by their use.

Invited Papers

1:20

2pNS2. Demonstrating compliance of noise exposure experienced by British military aircrew with noise legislation in the United Kingdom. Susan H. James (Aircrew Systems, QInetiQ, Rm. 2022 A5 Bldg QinetiQ, Cody Technol. Park, Farnborough, Hampshire GU14 0LX, United Kingdom, shjames@qinetiq.com)

In 2005, the Control of Noise at Work Regulations were introduced into UK law and imposed two Exposure Limits Values, one for continuous noise and one for impulse noise. The Ministry of Defence (MOD) has mandated that the legislation will apply in full throughout the military, and hence, there has been a requirement to understand the types of exposure military operators are exposed to. Military aircrew have historically been exposed to high levels of continuous noise and the more stringent legislation has necessitated enhanced noise mitigation, including the development of new hearing protection technologies. However, in more recent years, military aircrew have become increasingly exposed to weapon firing and the MOD are now addressing the implications of the combined effects of the two different exposures and their impact on compliance with the legislation. In-flight measurement of continuous and impulse noise exposure has been conducted in a range of aircraft and the two exposures assessed for compliance with the legislative criteria. To meet the protection requirements in all platforms, enhanced hearing protection technologies are being developed, including Adaptive Digital Active Noise Reduction and comms processing techniques.

1:40

2pNS3. Noise exposure criteria for continuous noise: A case for reducing the 8 hour open ear exposure level. Hilary Gallagher, Richard L. McKinley (Air Force Res. Lab., Battlespace Acoust. Branch, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, hilary.gallagher.1@us.af.mil), Melissa A. Theis (Air Force Res. Lab., ORISE, Wright-Patterson AFB, OH), and Elizabeth A. McKenna (Air Force Res. Lab., Henry M Jackson Foundation, Wright-Patterson AFB, OH)

Personnel working in hazardous noise environments can be protected from noise induced hearing loss (NIHL) or other hearing related disabilities when their exposures are limited. Limiting noise exposures can be accomplished by engineering controls, administrative controls, and/or with the use of personal protective equipment. Damage risk criteria (DRC) were developed as a guide to limit personnel noise exposure in order to reduce the risk of NIHL. With the exception of the US Occupational Safety and Health Administration

(OSHA), the US and many European countries accepted the DRC of 85 dB for 8 hours with a 3 dB per doubling exchange rate. Recent studies of the auditory response to noise dose, conducted at the Air Force Research Laboratory, found that A-weighted open ear exposures may be more hazardous (i.e., have a higher effective noise dose) than equal A-weighted level protected ear exposures. A potential conclusion of these studies is that hearing protectors are more effective than previously thought and reducing the open ear exposure criteria may be needed in order to minimize NIHL. This presentation will describe the pros and cons of reducing the 85 dBA DRC.

2:00

2pNS4. Calculating total daily noise exposure using aircrew flight equipment noise attenuation and fight segment noise data. Daniel A. Gross (US Navy, 48110 Shaw Rd., Bldg. 2187, Patuxent River, MD 20670, Daniel.A.Gross@navy.mil)

The workplace of military aircrew is louder and more varied than most industrial environments. In the recent revision of MIL-STD-1474E, Noise Limits, octave band level design limits for helicopters have been replaced by a broader requirement based on noise exposure at the ear. This change brings 1474E into alignment with DoD hearing conservation policies which require that total daily noise exposure does not exceed an Leq of 85 dBA during an 8 hour period. Although these polices also require use of hearing protection whenever levels exceed 85 dBA, there is currently little written guidance for how to use hearing protector attenuation data to estimate exposure at the ear over the course of a typical mission profile. This paper will explain how attenuation data for flight helmets and communication headsets has been used by the Naval Air Systems Command to estimate noise exposure for crew and passengers in various types of aircraft.

2:20

2pNS5. Effects of noise exposure in combination with exposures to JP-8 jet fuel. David R. Mattie (711 HPW/RHDJ, 711 HPW/RHDJ, 2729 R St., Wright-Patterson Air Force Base, OH 45433-5707, david.mattie@us.af.mil), Larry D. Fechter (Loma Linda Veterans Affairs Medical Ctr., Loma Linda, CA), Jeff W. Fisher (National Ctr. for Toxicological Res., Little Rock, AR), John E. Stubbs (509 MDOS/SGOJ, Wright-Patterson Air Force Base, OH), and O'neil W. Guthrie (Northern Arizona Univ., Flagstaff, AZ)

The objective was to evaluate the potency of JP-8 jet fuel to enhance noise-induced hearing loss using inhalation exposure to fuel and simultaneous exposure to noise with male & female Fischer 344 (F344) rats for 6 h/d, 5 d/wk for 4 wk (200, 750, or 1500 mg/m^3). Parallel groups of rats also received nondamaging noise (85 dB) in combination with fuel, noise alone (75, 85, or 95 dB), or no exposure to fuel or noise. Computer software was used to generate a pure and precisely filtered white noise file of one octave band, centered at 8 kHz. The filtered file was then played through electrodynamic shakers that induced vibration from the outside in the metal plenum at the bottom of each exposure chamber to produce noise. Significant concentration-related impairment of auditory function measured by distortion product otoacoustic emissions (DPOAE) and compound action potential (CAP) threshold was seen in rats exposed to combined JP-8 (1500 mg/m^3) plus noise (85 dB) with trends toward impairment at 750 mg/m^3 JP-8 plus noise (85 dB). JP-8 alone exerted no effect on auditory function. Noise was able to disrupt DPOAE and increase auditory thresholds when noise exposure was at 95 dB. Two additional studies with JP-8 (1000 mg/m^3) and noise (85 dB), one with F344 rats and one with Long Evans rats did not affect auditory function. However, a pilot assessment indicated a central auditory processing dysfunction (i.e., impaired brainstem encoding of stimulus intensity) among F344 and LE rats exposed to JP-8 alone and JP-8 with noise.

Contributed Paper

2:40

2pNS6. The effects of military training vibration on rock art. Timothy Lavallee (LPES, Inc., 14053 Lawnes Creek Rd., Smithfield, VA 23430, tlavallee@lpesinc.com) and Jannie Loubser (Stratum UnLtd. LLC, Alpharetta, GA)

This presentation will provide an overview of potential impacts of vibration due to military training on Native American rock art. Military training is often conducted in areas near culturally sensitive landmarks or structures, and information regarding the effects of vibration on these types of structures is limited. An extensive literature review was conducted to determine the current state of information regarding vibration effects on rock art and other cultural resources. The condition of rock art can vary from the extremely stable to fragile; and based on the existing studies, thresholds for effects were defined in term of stability class for different rock art structures. Vibrations and peak sound levels from maneuver and live fire training activities were estimated and compared to these effects thresholds. Both airborne and groundborne vibrations decreased with distance and at some point attenuate below the threshold of adverse effects. Critical distances for effects were estimated for a variety of training activities—including demolition activities, ground maneuvers, mortar fire, and rotorcraft operations. Results will be provided in both peak particle velocity and peak sound levels. These results as well as potential best management practices will be discussed.

Session 2pPA

Physical Acoustics: Acoustic Characterization of Critical Phenomena

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Veerle M. Keppens, Cochair
Univ. of Tennessee, Dept. Materials Science and Engineering, Knoxville, TN 37996

Chair's Introduction—1:10

Invited Papers

1:15

2pPA1. Variations in elastic and anelastic properties as indicators of static and dynamic strain relaxation phenomena associated with phase transitions. Michael A. Carpenter (Dept. of Earth Sci., Univ. of Cambridge, Downing St., Cambridge CB2 3EQ, United Kingdom, mc43@esc.cam.ac.uk)

Almost all phase transitions are accompanied by some degree of lattice strain, typically varying between a few per mil to a few per cent, but extending to 5–10% for martensitic transitions. It is inevitable, therefore, that there will also be changes in elastic properties and, because the elastic moduli are susceptibilities, these are typically in the range of 10's of per cent. At the same time, the associated transformation microstructures may be mobile under the action of external stress and therefore give rise to anelastic losses in a dynamical mechanical measurement. It has turned out that Resonant Ultrasound Spectroscopy, in a frequency window ~0.1–2 MHz, is a particularly powerful method for characterizing elastic and anelastic behavior as functions of temperature and magnetic field for ferroic and multiferroic phase transitions in a wide range of materials. Examples of recent collaborative studies, relating to magnetoelastic effects in $EuTiO_3$, multiferroic transitions in $Pb(Fe_{0.5}Nb_{0.5})O_3$, the influence of grain size on Jahn-Teller $\times$ charge ordering in $La_{0.5}Ca_{0.5}MnO_3$, ferroelectric and ferroelastic transitions in metal organic frameworks, and martensitic transitions in Heusler alloys, will be used to demonstrate specific mechanisms and kinetics of static and dynamic strain relaxation phenomena.

1:45

2pPA2. A key test of the theory of critical phenomena using acoustics in liquid helium. Julian D. Maynard (Physics, Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

A significant success of modern theoretical physics has been in the study of critical phenomena, where a system displays singular properties while undergoing a transition between different phases of matter. The modern theory (recognized by a Wolf Prize and a Nobel Prize) involves the notions of length scale invariance, power law singularities, the renormalization group etc. A significant experimental test of the theory has been the study of the transition between normal liquid and superfluid behavior of liquid helium-4 (recognized with a London Prize). The singular properties which characterize a critical transition typically involve a divergence in a susceptibility, such as a heat capacity, compressibility, or other property which may couple to a local configuration of atoms or molecules. Since these latter properties may be probed with sound waves, acoustics can be a useful probe of critical behavior. In the case of superfluid helium there are five fundamentally different modes of sound propagation, and this acoustic abundance contributed to the success of the critical phenomenon theory test. This talk will review the theory of critical phenomena, describe the different modes of sound propagation in superfluid helium, and summarize the results of the key test of the critical phenomenon theory.

2:15

2pPA3. Structural and magnetic phase transitions in $EuTi_{1-x}Nb_xO_3$: A resonant ultrasound spectroscopy study. Ling Li (Mater. Sci. and Eng., Univ. of Tennessee, 1508 Middle Dr., Knoxville, TN 37909-2879, liling822@gmail.com), James R. Morris (BES Mater. Sci. and Eng. Program, Oak Ridge National Lab., Knoxville, TN), Michael R. Koehler (Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN), Zhiling Dun, Haidong Zhou (Phys. and Astronomy, Univ. of Tennessee, Knoxville, TN), Jiaqiang Yan, David G. Mandrus, and Veerle M. Keppens (Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN)

We have investigated the structural and magnetic phase transitions in $EuTi_{1-x}Nb_xO_3$ ($0 \leq x \leq 0.3$) with synchrotron powder X-ray diffraction (XRD), resonant ultrasound spectroscopy (RUS), and magnetization measurements. The Pm-3m $\leftrightarrow$ I4/mcm structural transition in pure and doped compounds is marked by a pronounced step-like softening of the elastic moduli near T_S, which resembles that of $SrTiO_3$ and can be adequately modeled using the Landau free energy model employing the same coupling between strain and octahedral tilting order parameter as previously used to model $SrTiO_3$. Upon Nb doping, the cubic-to-tetragonal structural transition shifts to higher temperatures and the room temperature lattice parameter increases while the magnitude of the octahedral tilting decreases. In addition, Nb substitution for Ti destabilizes the antiferromagnetic ground state of the parent compound and long range ferromagnetic order is observed in the samples with $x \geq 0.1$.

2:45–3:15 Break

3:15

2pPA4. Fluctuation-driven attenuation and dispersion of sound near the critical points of fluids. Keith A. Gillis and Michael R. Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov)

We discuss the extraordinary growth in the attenuation of sound α_λ and in the dispersion in the speed of sound c, which occurs near all liquid-vapor critical points. The attenuation and dispersion have been measured over 5 orders of magnitude in frequency. Remarkably, the data collapse onto universal, theoretically predicted curves. The theory considers equilibrium density fluctuations near the critical point where the fluctuations are large compared with the particle spacing. These density fluctuations have a distribution of sizes characterized by the correlation length ξ and a distribution of lifetimes characterized by the relaxation time τ. As the critical point is approached, ξ and τ diverge with the universal power laws $\xi \propto r^{-0.63}$ and $\tau \propto r^{-1.93}$, where r is a measure of the distance from the critical point. [At the critical density ρ_c, $r \equiv (T - T_c)/T_c$.] In the low-frequency limit ($\omega\tau \ll 1$), the attenuation grows as $\alpha_\lambda \propto r^{-1.93}$ and the speed of sound approaches zero as $c \propto r^{0.055}$. When $\omega\tau \gg 1$, α_λ approaches a maximum, non-universal limit and strong dispersion is present. Low-frequency sound waves reach the condition $\omega\tau = 1$ closer to T_c and deeper into the asymptotic critical regime than do high-frequency sound waves. In Earth's gravity, we show that stirring a near-critical fluid reduces stratification and enables measurements closer to the critical point. We compare the attenuation and dispersion for pure fluids near the liquid-vapor critical point with that of binary liquid mixtures near the consolute point.

Contributed Papers

3:45

2pPA5. Simple homodyne ultrasound interferometer for solid state physical acoustics. Oleksiy Svitelskiy, David Lee (Physics, Gordon College, 255 Grapevine Rd., Gordon College, Wenham, MA 01984, oleksiy.svitelskiy@gordon.edu), John Grossmann (Columbia Univ., New York, NY), Lynn A. Boatner (Oak Ridge National Lab., Oak Ridge, TN), Grace J. Yong (Towson Univ., Baltimore, MD), and Alexey Suslov (National High Magnetic Field Lab., Tallahassee, FL)

Ultrasonic pulse-echo technique is a valuable and non-destructive tool to explore elastic properties of materials. We propose a new instrument based on mass-produced microchips. In our design, the signal is processed by an AD8302 RF gain and phase detector (www.analog.com). Its phase output is linearly proportional to the phase difference between the exciting and response signals. The gain output is proportional to the log of the ratio of amplitudes of the received to the exciting signals. To exclude the non-linear fragments and to enable exploring large phase changes, we employ a parallel connection of two detectors, fed by in-phase and quadrature reference signals, respectively. The interferometer was tested by measuring the temperature dependences of both sound speed and attenuation in metallic glasses as well as in ferroelectric $KTaNbO_3$ (KTN) single crystals. The instrument allows for exploring phase transitions with precision of $\Delta V/V \sim 10^{-7}$ (V is ultrasound speed) in the broad dynamic range from -60 to $\times 20$ dBm. These qualities allowed us to detect the theoretically predicted, but not observed previously velocity kink at the KTN phase transition from tetragonal to orthorhombic symmetry, whereas the attenuation curve showed new features in the development of the low-temperature structure of the KTN crystal.

4:00

2pPA6. Temperature and pressure effects on elastic properties of lead magnesium niobate-lead titanate relaxorferroelectric material. Sumudu P. Tennakoon and Joseph R. Gladden (Phys. and Astronomy & NCPA, Univ. of MS, 145 Hill Dr., NCPA, Rm. 1077, University, MS 38677, sptennak@go.olemiss.edu)

Lead magnesium niobate—lead titanate (PMN-PT) exhibits exceptional electromechanical properties, considered as a highly efficient transduction material for vibration energy harvesting and acoustic sensing applications. It is reported in the literature that the PMN-PT undergoes structural phase transitions with changes in temperature and the chemical composition. We seek to gain insight into the phase diagram of PMN-PT using temperature and pressure dependence of the elastic properties. Single crystal PMN-PT with chemical composition close to the morphotropic phase boundary (MPB) was used in a resonant ultrasound spectroscopy (RUS) study performed in the temperature range from room temperature to 773 K and the pressure range from near vacuum to 3.4 MPa. At atmospheric pressure, significantly high acoustic attenuation of the PMN-PT material is observed at the temperatures below 400 K. Strong stiffening is observed in the temperature range of 400 K–673 K, followed by a gradual softening at higher temperatures. With the varying pressure, we observed an increased pressure sensitivity of elastic properties of the PMN-PT material that can be localized to the temperature regime where the strong stiffening is observed. As time allows, the behavior of PMN-PT upon cooling below room temperature will also be discussed.

4:15

2pPA7. Acousto-optic investigation of acoustic anisotropy in paratellurite crystals. Farkhad R. Akhmedzhanov and Ulugbek Saidvaliev (Samarkand State Univ., 15 University Blvd., Samarkand 140104, Uzbekistan, farkhad2@yahoo.com)

Propagation velocity and attenuation coefficient of acoustics waves in paratellurite crystals were measured by Bragg diffraction of light in the frequency range of 0.4–1.6 GHz. These results were used for calculation of real and imaginary components of the complex tensor of elastic constants. The analysis of the anisotropy of attenuation was carried out for the acoustic waves of different polarization propagating in the crystallographic planes, which are orthogonal to symmetry axes of second and forth order. The strongest anisotropy of acoustic attenuation and phase velocity is observed for the transverse acoustic waves propagating in the plane (1-10). It is shown that the attenuation reduces the integral efficiency of diffraction in several times in the acousto-optic deflector in which is used an oblique cut of paratellurite and the transverse acoustic wave propagating at an angle of 6 degrees to the [110] axis in the plane (1-10).

Session 2pSA

Structural Acoustics and Vibration, Noise, and Architectural Acoustics: Structural Acoustics and Vibration in Buildings

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Benjamin Shafer, Cochair
Technical Services, PABCO Gypsum, 3905 N 10th St, Tacoma, WA 98406

Chair's Introduction—1:00

Invited Papers

1:05

2pSA1. A preview of the second edition of the AISC Design Guide 11 "Vibrations of Steel Framed Structural Systems due to Human Activity". Eric E. Ungar (Acentech, 15 Considine Rd., Newton Ctr., MA 02459, eungar@acentech.com)

The Steel Design Guide 11, "Floor Vibrations due to Human Activity," which originally was published in 1997, has been refurbished, updated, and broadened. At the time of the writing of this abstract, a draft is being reviewed by its sponsor, the American Institute of Steel Construction. Like the original, the second edition presents easily used means for predicting the vibrations of floors of steel construction due to typical walking and for assessing the acceptability of these vibrations in relation to human comfort. The new edition relies on more recent data on footfall forces and includes prediction methods (analytically derived and empirically adjusted on the basis of comparison with experimental data) for evaluating the acceptability of walking-induced vibrations on equipment and facilities whose limits are expressed in terms of a variety of measures. Analyses of stairs and pedestrian bridges are included, and modeling of structures with steel joists is discussed. An extensive chapter is devoted to the application of finite-element analysis to floor structures.

1:30

2pSA2. The relationship of vibration and groundborne/structureborne noise in buildings relative to vibration in the ground from exterior sources. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Ground vibration generated by roadway traffic, rail, and transit systems can be transmitted into buildings were it can either be perceived as feelable vibration or as sound radiated from the interior building surfaces, i.e., floors, walls, and ceilings. The amount of vibration that is transmitted into a building is affected by a reduction as the vibration is transmitted from the ground to the foundation of the building (i.e., coupling loss) and amplification due to resonances within the building structure. This paper will discuss these effects and measurements that have been conducted to quantify the effects for common residential building construction adjacent to rail lines. A brief discussion regarding the relationship of groundborne/structureborne noise to interior vibration from exterior sources of ground vibration will be included.

1:55

2pSA3. Control of amplification of ground vibration in joist-framed buildings. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

For multifamily buildings near railroads, it is possible to directly measure the ground vibration caused by passing trains. The indoor vibration levels are determined by predictions of the transfer of vibration from ground to foundation, propagation up the building, and amplification by floor resonances. Large amounts of amplification have been observed in lightweight joist-framed structures. One mitigation method is to stiffen the structural system, both to raise the resonant frequency to a less objectionable range, and to reduce the magnitude of the resultant vibration. However, the benefit of this mitigation has rarely been quantified or presented. Measurements were performed in a multifamily building adjacent to railroad tracks that had been stiffened over a portion of the building. Vibration propagation and amplification were measured in the stiffened and non-stiffened portions, and the effect of mitigation is evaluated in the context of anticipated results and actual measured levels.

Contributed Papers

2:20

2pSA4. Modeling, designing, and testing low frequency impact isolation solutions for building structures with fitness centers above grade. Norman D. Varney (Kinetics Noise Control, 6300 Irelan Pl., Dublin, OH 43017, nvarney@kineticsnoise.com)

There is a trend for fitness centers to find a quick, inexpensive occupancy and ideal location in a vacant office or retail space, often above grade. The tenants sharing the building structure are then subjected to sudden airborne noise and structureborne vibration from the impact energy generated by heavy weights being dropped onto the floor. Because of this trend there is a need to analyze the large, low frequency, shock generated energies that travel through the structure. This paper will discuss the process of designing an experimental floor assembly and apparatus to test weight drops, developing a modeling program to simulate weight drops using three degrees of freedom, which can be tailored to varying isolation solutions, and the means of physically testing the results.

2:35

2pSA5. Noise control within building water supply lines. Elliott Gruber and Kenneth Cunefare (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, e_gruber@gatech.edu)

Successful treatment of fluid-borne noise in a building's plumbing system reduces noise propagation throughout the building, which improves occupant comfort and extends component lifetimes. Circulation pumps and quick-closing valves contribute to noise in water systems. Pump noise is generally untreated. Quick-closing valves m may cause water hammer, which can be treated with a water hammer arrestor (WHA). Common WHA's functions by adding a compliant gaseous volume to the system; the gas volume is sealed from the system by a free piston. Control of both water-borne noise and water hammer may be achieved by a flow-through device integrating a compliant, voided polymer. The performance of current WHAs diminish over time as (1) mineral deposits degrade sealing effectiveness and (2) the gas permeates the seals; a voided polymer WHA will not suffer from these drawbacks. Prior work has demonstrated that a voided polymer is an effective source of compliance for noise control in oil systems at the anticipated pressure and temperature range of water systems. Furthermore, the acoustic impedance of oil is similar to water. Basic modeling and performance data for a prototype device will be presented.

TUESDAY AFTERNOON, 3 NOVEMBER 2015 GRAND BALLROOM 6, 1:30 P.M. TO 3:00 P.M.

Session 2pSCa

Speech Communication: Forensic Acoustics, Speaker Identification, and Voice

Robert A. Fox, Chair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002

Contributed Papers

1:30

2pSCa1. Reconsideration of forensic acoustics. Harry Hollien (Linguist & IASCP, Univ. of Florida, IASCP, Univ. Fla, Gainesville, FL 32611, hollien@ufl.edu)

The high talent level of in our Society members is unquestioned. However, it appears that some of us have not yet reached full maturity relative to Forensic Acoustics. While the adversarial aspect of Forensics does not parallel our investigational rigor, the number of variables they face is much greater than ours. Accordingly, development of acoustic systems for forensic use requires different approaches than we typically use. Speaker identification (SI) constitutes a good example here. Fifty years ago, it was agreed that a valid SI system could be quickly created but that has not happened. In response, the undersigned adapted the scientific method (and its standards) as a model for SI system development. The resulting research was modestly successful but the approach was not considered "useful". Later, the Daubert test confirmed this model (but ordinarily was circumvented). Now, the National Academy of Science has verified these models and is insisting on proper developmental protocol. It is no longer permissible to develop an algorithm-based device, and apply it, without proper demonstration of its validity. The impact of these research-based standards—now required for creating Forensic systems—will be discussed, as will possible responses.

1:45

2pSCa2. Voice disguise and speaker identification. Grace S. Didla (Post Doctoral Researcher, Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, didlagrace@gmail.com) and Dr. Harry Hollien (Professor Emeritus, Linguist, Univ. of Florida, Gainesville, FL)

Voice disguise involves deliberately changing one's voice to conceal their identity. It is most often associated with crimes such as kidnapping, fraud, etc. In addition to being a general problem in the world of crime, it is a particular one in the area of Speaker Identification (SI) and for the following reasons: **(1)** There appears to be no limit on human ingenuity in the types of disguise that can be employed. **(2)** While there are unlimited ways to disguise speech, there seem to be far too few studies to provide useful information about it. **(3)** Further these studies are narrow in scope. This can be attributed to the extensive number of variables (type of disguise, familiarity with the speaker, language, dialect, etc.) and the need for their control. **(4)** Because of these limitations, a great deal more needs to be known about the major types of disguise and the ones detrimental to SI. Thus, the numerous issues plaguing voice disguise must be addressed systematically through an integrated research program. A review of their nature, the research needed and the reasons for conducting the program will be discussed. Data from a sample experiment of this type will be presented.

2:00

2pSCa3. Vibrational and acoustic consequences of changes in subglottal pressure, vocal fold stiffness, vocal fold approximation, and vocal fold thickness. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Using a three-dimensional continuum model of phonation, this study investigates the effects of changes in subglottal pressure, vocal fold approximation and stiffening, and vocal fold medial surface thickness on vocal fold vibration and acoustics. The results show that increasing subglottal pressure leads to more or less uniform increase in harmonic energy across a very large frequency range as well as significant increase in noise production. Reduced noise production can be achieved by increasing medial surface thickness, vocal fold approximation, or vocal fold stiffening. Increasing vocal fold thickness and stiffness also lead to increased production of high-frequency harmonics. The closed quotient of vocal fold vibration depends primarily on the medial surface thickness of the vocal fold, with the closed quotient increasing with increasing thickness. The closed quotient also slightly decreases with increasing subglottal pressure, but increases with increasing vocal fold approximation and stiffening. These results suggest that, in addition to increasing subglottal pressure, vocal loudness can be also increased by increasing vocal fold thickness, approximation and/or stiffening to increase production of higher-order harmonics and reduce noise production, which leads to a perceived increase in vocal intensity. [Work supported by NIH.]

2:15

2pSCa4. Behavioral and computational estimates of breathiness and roughness over a wide range of dysphonic severity. David A. Eddins, Arianna Vera-Rodriguez, Mark D. Skowronski (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu), and Rahul Shrivastav (Univ. of Georgia, Athens, GA)

Perceptual evaluations of dysphonic voices frequently involve evaluation of the breathy and rough qualities. It is important to know how well one can disambiguate such VQ percepts in a perceptual evaluation so that accurate assessments can be made, focused treatment targets can be set, and outcomes can be quantified. In this study, 10 voices that varied over a wide range of breathiness and 10 voices that varied over wide range in roughness were selected from the University of Florida Disordered Voice Database (UFDVD). A single-variable matching task designed specifically for breathiness or roughness evaluation was used to index the perceived breathiness and roughness of the set of 20 stimuli. Because computational estimates of pitch salience (pitch strength) have been strongly associated with perceived breathiness, we also included a perceptual evaluation of pitch strength using an anchored magnitude estimation task. In an effort to better understand the interactions among perceived roughness and breathiness, a series of computational algorithms was used to estimate perceived breathiness and perceived roughness. Together, these results will help us understand the natural covariance of voice qualities and the ability to evaluate separate voice qualities independently.

2:30

2pSCa5. The acoustics of perceived creaky voice in American English. Sameer ud Dowla Khan, Kara Becker (Linguist, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameeruddowlakhan@gmail.com), and Lal Zimman (Linguist, Univ. of California Santa Barbara, Santa Barbara, CA)

We compared auditory impressions of creaky voice in English to acoustic measures identified as correlates of contrastive voice qualities in other languages (e.g., Khmer, Chong, Zapotec, Gujarati, Hmong, Trique, and Yi). Sixteen trained linguistics undergraduates listened to the IP-final word 'bows' produced five times each by five American English speakers reading the Rainbow Passage, and gave a rating from 0 (no creak) to 5 (very creaky). Results show that stronger auditory impressions of creak are significantly correlated with lower f0, lower cepstral peak prominence (CPP), lower harmonics-to-noise ratios (HNR), and higher subharmonics-to-harmonics ratio (SHR). This suggests that listeners perceive greater creakiness as the voice becomes lower pitched, less periodic, and more audibly interspersed with subharmonic frequencies (i.e., diplophonia). Notably, none of the spectral amplitude measures proposed as acoustic correlates of glottal configurations for creaky voice in other languages (e.g., lower H1-H2 for smaller open quotient, lower H1-A1 for smaller posterior aperture, lower H1-A3 for more abrupt closure, etc.) was significantly correlated with these judgments in any expected direction. Taken together, these results suggest that while listeners consistently use pitch and periodicity as cues to creak, speakers might be varying in their articulatory strategies to achieve those acoustic effects.

2:45

2pSCa6. Evaluating acoustic measurements of creaky voice: A Vietnamese case study. Nadya Pincus, Angeliki Athanasopoulou, Taylor L. Miller, and Irene Vogel (Linguist & Cognit. Sci., Univ. of Delaware, 125 E Main St., Newark, DE 19716, npincus@udel.edu)

It has been proposed that there are two broad categories of creaky voice (CV), laryngealized and aperiodic. Moreover, several subdivisions have been proposed for both categories (Keating & Garellek, 2015), and various combinations of acoustic properties have been associated with each. It remains unclear, however, how to determine which type of CV a language has and which acoustic measurements to rely on. We address this problem with two rising tones in Vietnamese differing in phonation. All of the phonation measurements we tested with ANOVA were statistically significant ($p < .01$) in the distinction between the two tones, and thus inconclusive as to the type of CV. We propose that an additional binary logistic regression analysis be applied to the various measurements to determine the extent to which each one contributes to classifying creaky vs. modal vowels; and this, in turn, can inform us about the nature of the CV in the language. Specifically in Vietnameses, we found that HNR yields the strongest classification result (84%); the others were closer to chance (58–68%). We can thus conclude that the CV used in Vietnamese is the aperiodic type, as evidenced by the role of irregular F0 as opposed to the other phonation properties.

Session 2pSCb

Speech Communication: Speech Perception Potpourri (Poster Session)

Tuli Morrill, Chair
Michigan State University, 4400 University Drive, 3E4, Fairfax, VA 22030

Authors will be at their posters from 3:30 p.m. to 5:00 p.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 1:00 p.m. to 5:00 p.m.

Contributed Papers

2pSCb1. Detailing vowel development in infancy using cortical auditory evoked potentials and multidimensional scaling. Kathleen McCarthy (Speech, Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, kathleen.mccarthy@ucl.ac.uk), Katrin Skoruppa (Universität Basel, Basel, Switzerland), and Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, London, United Kingdom)

The present study used an efficient measure of perceptual sensitivity to map perception across the British English vowel space for 80 monolingual English infants (4–5, 8–9, and 10–11 months old). Auditory evoked potentials were measured for spectral changes between concatenated vowels, which, for infants, typically evokes a positivity about 150–200 ms after each spectral change. These were measured for 28 pairs of seven monophthongal vowels (/i/, /ɪ/, /ɛ/, /a/, /ɑ/, /ɔ/, /u/) that were presented in a random concatenated sequence with changes every 300–400 ms. ERPs were averaged across epochs following each spectral change, with the magnitude of the response for each vowel pair used as similarity measure for multidimensional scaling. The 4–5 month old infants had two-dimensional perceptual maps that closely matched the F1 and F2 acoustic differences between vowels. In contrast, the older infant response was less related to acoustic differences and they had selectively larger responses for neighbors around the vowel quadrilateral (e.g., /i/-/ɪ/); suggesting a shift to a more phonologically driven processing. These results provide a more detailed picture of phonetic development than has been shown before, and demonstrate an efficient procedure to map speech processing in infancy.

2pSCb2. Speech perception capabilities in children several years after initial diagnosis of auditory processing disorders. Rachel Crum (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 125 East Main St., Newark, DE 19716, rachelc@udel.edu), Jennifer Padilla (Dept. of Psychol. and Brain Sci., Univ. of Delaware, Newark, DE), Thierry Morlet (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE), L. A. Greenwood (Pediatrix Audiol. Services, Falls Church, VA), Jessica Loson, Sarah Zavala (Audiol., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE), and Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE)

This study focuses on the progress of speech perception capabilities in children with auditory processing disorder (APD) measured several years after the initial diagnosis. A recent longitudinal study showed listening and communication difficulties of children diagnosed with APD persist into adulthood (Del Zoppo, Sanchez, and Lind, 2015). In the current study, we examined the speech perception progress of 21 children selected from the Auditory Processing Disorder database of 255 school-aged children in the Mid-Atlantic region as having an initial diagnosis of APD and an APD reassessment at our facility several years later. Average age for initial assessment was 7.9 (7–10) years with the most recent APD evaluation ages 11.5 (9–16) years. Results show that 81% of the children still had auditory processing deficits in their most recent evaluations, 9% had only associated area issues, and 10% exhibited typical auditory processing performance. We found that average standardized scores for all tests, except auditory figure ground, collectively increased in the second assessment. These tests include: competing words, competing sentences, filtered words, composite scores, and phonemic synthesis. While the auditory processing skills of some children with APD are improving over time, some children still show impairment in several processing areas.

2pSCb3. Adults' perceptual voicing boundaries of 2-year-olds' citation form speech. Elaine R. Hitchcock (Commun. Sci. and Disord., Montclair State Univ., 116 West End Ave., Pompton Plains, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Haskins Labs., New Haven, CT)

One way that speakers distinguish between phonemic categories is through voicing, frequently measured using voice onset time (VOT). Much perceptual research on voicing identification and discrimination has used synthetic speech stimuli varying in VOT. Results from adult listeners typically show stable crossover regions in the 20–35 ms range. Subsequent work, however, reveals that listeners' VOT boundaries vary with speech rate; further, an extensive history of research into vowel perception indicates that listeners normalize across vocal tract sizes. These considerations lead to the possibility that adult voicing boundaries may differ between the speech of adults vs. children, since children have slower speech rates and smaller vocal tracts. The present study obtained adult discrimination data for natural productions of bilabial and alveolar cognate pairs produced by 2–3-year-old monolingual, English-speaking children. Randomized stimuli were presented twice to 20 listeners resulting in 4,000 rated stimuli per category. The findings show 50% crossover points for VOT values at 28 ms for bilabials and 32 ms for alveolar phonemes. Such outcomes are consistent with past work based on adult data and suggest that mature listeners do not use substantially different perceptual criteria for judging voicing in children's speech. **Declaration of Interest Statement** The authors have no financial or nonfinancial disclosures to report.

2pSCb4. Top-down influences in perception with spectrally degraded resynthesized natural speech. Jane Smart and Stefan Frisch (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD 1017, Tampa, FL 33620, jbsmart@mail.usf.edu)

Lexical processes can influence perception of ambiguous phonemes (e.g., Ganong, 1980). To date, research has focused on these influences in quiet conditions with stimuli that have not been degraded. This project examines the interplay between lexical and acoustic information in speech perception, with stimuli in non-degraded and spectrally degraded conditions. Continua of /t, k/ onsets were developed by wavelet resynthesis of natural speech using TandemSTRAIGHT software, concatenated to the vowel × coda portion of words/nonwords, and distorted using AngelSim (TigerCIS) vocoding software. Normal hearing adult participants identified the onset phonemes in non-degraded and spectrally degraded conditions. Identification functions and the effects of lexical status in phoneme perception with spectrally degraded stimuli will be discussed.

2pSCb5. Brain activity for sound symbolism in visual size judgment: Combinations of voiced and voiceless plosives with a vowel "o" or "i". Sachi Itagaki, Shota Murai, Shizuko Hiryu, Kohta I. Kobayasi (Graduate School of Life and Medical Sci., Doshisha, 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 6100394, Japan, dmp1006@mail4.doshisha.ac.jp), Jan Auracher (Konan, Kobe, Japan), and Hiroshi Riquimaroux (Graduate school of Life and Medical Sci., Doshisha, Kyoto, Japan)

Sound symbolism is an idea that sounds itself has the impression. In most of the previous psychology and linguistics researches, stimuli were presented visually with alphabets, and subjects directly answered the impression of the sound. The purpose of this study is that establishing a behavioral paradigm applicable to functional magnetic resonance imaging (fMRI) research when the sound stimulus was presented aurally. In this experiment, we focused on sound symbolism in visual size. Subjects were required to answer visual size difference between standard and target stimulus. Visual stimuli were a gray circle on black background LCD screen. Sound stimuli were /bobo/ and /pipi/, and were assumed to have impression of "bigger" and "smaller," respectively, according to previous researches. Currently, brain activity of the sound symbolism is examined in MRI using ours previous behavior paradigm. The result suggested that our paradigm is able to use for fMRI study. However, MRI imaging results shows that the location and amount of activity is different by subjects, suggesting the neural substrate of the sound symbolism could vary between individuals. Relationships between the brain differences and individual behavioral differences will be discussed.

2pSCb6. Perceptual load of divided attention modulates auditory-motor integration of voice control. Hanjun Liu, Ying Liu, and Zhiqiang Guo (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

The present study sought to examine how the auditory-motor processing of feedback errors during vocal pitch regulation is modulated as a function of divided attention. Subjects were exposed to pitch perturbations in voice auditory feedback and flashing lights on the computer screen, during which they were asked to divide their attention to both auditory and visual stimuli by counting the number of both of them. The presentation rate of the visual stimuli (inter-stimulus intervel) was manipulated to produce a low, intermediate, and high attentional load. The results revealed that, as compared to the low and intermediate attentional load conditions, the high attentional load condition elicited significantly smaller magnitudes of vocal responses but larger amplitudes of N1 and P2 responses to pitch feedback perturbations. These findings demonstrate the effect of attentional load on the auditory-motor processing of pitch feedback errors during divided attention, suggesting that perceptual load of visual attention interferes the attentional modulation of auditory-motor integration in voice control.

2pSCb7. Speaker identification using auditory modeling and vector quantization. Konstantina Iliadi and Stefan Bleeck (Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, konstiliadi@gmail.com)

Speaker identification (SID) aims to identify the underlying speaker(s) given a speech utterance. SID systems can perform well under matched training and test conditions but their performance degrades significantly because of the mismatch caused by background noise in real-world environments. Achieving robustness to the SID systems depends very much on the front-end (or feature extractor), which is the first component in an automatic speaker recognition system. Feature extraction transforms the speech signal into a compact representation that is more discriminative than the original signal. We present on our poster a new system where the parametrization of the speech is based on an auditory model called Auditory Image Model (AIM). Two experiments were performed for two different sets of speakers. Experiment 1 identified the most informative regions of the auditory image that can indicate speaker recognition. Experiment 2 consisted of training 10 and 60 speakers using clean speech and testing those two groups using speech in the presence of babble noise of eight speakers for 5 SNRs. The results suggest that the extracted auditory feature vectors led to much better performance, i.e., higher SID accuracy, compared to the MFCC-based recognition system especially for low SNRs.

2pSCb8. Digit recognition with phonological features. Vipul Arora, Aditi Lahiri (Faculty of Linguist, Philology & Phonet., Univ. of Oxford, University of Oxford, Oxford OX1 2HG, United Kingdom, vipul.arora@ling-phil.ox.ac.uk), and Henning Reetz (Dpt. of Empirical Linguist, Goethe-Univ. Frankfurt, Frankfurt, Germany)

Our project aims to identify English digits, employing an approach to speech recognition built on principles derived from phonological knowledge and neurolinguistic experiments on how the humans perceive and process speech. This is in contrast to current ASR systems, which rely upon statistical machine learning using thousands of hours of speech training data and vast amount of computations, with little reliance on any aspect of phonological features. We focus on digits to test our approach with a circumscribed set of elements, which have considerably different phonemes (e.g., the sounds in 'five', 'two' and 'six' are all different). Our system converts the acoustic signal to a set of phonological features whose combinations are used to access words. The features are speaker and language independent, with the intention of building a system easily adaptable to other languages without re-training the acoustic front-end. A number of acoustic parameters (e.g, LPC, spectral and energy differences) were investigated and their relative importance compared for a robust estimation of phonological features; e.g., spectral slope below and above 2500 Hz disambiguates sibilants. However, distinguishing sonorant sounds requires a range of other parameters. We tested our system on the American English TIDIGIT database achieving unexpected success rates.

2pSCb9. Links between the perception of speaker age and sex in children's voices. Peter F. Assmann, Michelle R. Kapolowicz, David A. Massey (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda (Dept. of Linguist, Univ. of California, Davis, Davis, CA), and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

At a recent meeting [Assmann *et al., J. Acoust. Soc. Am.* **135**, 2424 (2014)] we reported two experiments on the perception of speaker age and sex in children's voices, along with two models to predict listeners' judgments. The stimuli were vocoded /hVd/ syllables produced by 140 speakers, ages 5 through 18, processed to simulate a change in the sex of the speaker. Experimental conditions involved swapping the fundamental frequency (F0) contour and/or the formant frequencies (FF) to the opposite-sex average within each age group. The present study extended the original experiments by requiring each listener to judge both age and sex on each trial to investigate the relationship between the two perceptual responses. Results revealed that age estimation error is systematically linked to sex misclassification, particularly in older children. In the unswapped condition, age estimates tended to be lower if the voice was identified as male, relative to the same voice heard as female. The condition with both F0 and FF swapped approached the opposite pattern of results; however, the remaining discrepancy indicates these are not the only cues for the perception of age and sex in children's voices.

2pSCb10. Understanding southern British, Glaswegian, and Spanish English accents: Speech in noise and evoked potentials. Petra Hödl, Melanie Pinet, Bronwen Evans, and Paul Iverson (Univ. College London, University College London, 2 Wakefield St, London, United Kingdom, p.hodl@ucl.ac.uk)

Speech recognition in noise is affected by the accents of the speakers and listeners, but it is not clear how overall accuracy is linked to the underlying perceptual and lexical processes. The present study investigated speech recognition for two native-accent groups (Southern British English and Glaswegian) and one non-native group (Spanish learners of English). Listeners were tested behaviorally on speech-recognition in noise, and using EEG measures of vowel perception (cortical evoked potentials to vowel spectra change) and lexical processing (N400). As expected, southern British English listeners were most accurate for southern British speech, Glaswegians were accurate for both Glaswegian and southern British English speech, and Spanish speakers had particular difficulty with Glaswegian. The EEG results demonstrated differences between groups in terms of both vowel and lexical processing. In particular, Glaswegian listeners differed in their lexical processing for the two native accents despite having similar speech-in-noise accuracy, and Spanish speakers appeared to use contextual information less than the other two groups. The results begin to demonstrate how differences at a perceptual level can be compensated for during lexical processing, in ways that are not apparent purely from recognition accuracy scores.

2pSCb11. Effect of menstrual phase on dichotic listening. Richard J. Morris and Alissa N. Smith (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu)

Women using birth control and those not using it completing weekly dichotic listening sessions for nine weeks. Results indicated no differences between the two groups for right ear advantage.

2pSCb12. Listener characteristics and the perception of speech in noise. Noah Silbert and Lina Motlagh Zadeh (Commun. Sci. and Disord., Univ. of Cincinnati, 3239 Bishop St. Apt #4, Cincinnati, OH 45220, motlagla@mail.uc.edu)

Speech communication is often made difficult by the presence of background noise. Much research on the perception of noise-masked speech has focused on the masking of phonetic information by different types of noise (e.g., white noise, speech-shaped noise, temporally modulated noise, multi-talker babble). The present work focuses on the relationships between some cognitive characteristics of listeners and accuracy in the identification of noise-masked consonants. 37 listeners identified numerous tokens of each of 4 consonants (p, b, f, v) in CV syllables produced by 8 talkers (4 male, 4 female) masked by 10-talker babble. Listeners also completed a number of tasks designed to measure selective attention: two dichotic listening tasks and two non-speech discrimination tasks. On each trial of the dichotic listening tasks, one or the other ear was cued visually (i.e., "right ear" or "left ear"), after which the listener indicated the talker sex or the consonant in the target ear, depending on task. In the two non-speech tasks, listeners discriminated either the frequency or the duration of broadband target noise bursts embedded in temporally modulated background noise. Analyses indicate a positive relationship between noise-masked speech accuracy and performance on the dichotic consonant identification and complex non-speech discrimination tasks.

2pSCb13. The effect of semantic cues on intelligibility: A comparison between spectrally sparse speech and natural speech in noise. Bahar Shahsavarani, Thomas Carrell (Commun. Disord., Univ. of Nebraska - Lincoln, 352 Barkley Memorial Ctr., Lincoln, NE 68583, sshahsav@cse.unl.edu), and Ashok Samal (Comput. Sci., Univ. of Nebraska - Lincoln, Lincoln, NE)

Listeners are known to make use of contextual cues when perceiving speech. Using contextual information is even more important when listening to distorted or difficult speech signals. For example, listeners benefit from contextual cues to compensate for the absence of fine acoustic-phonetic information when listening to natural speech in noise. In the present investigation, the effect of context was compared between four- and eight-channel spectrally degraded speech (Shannon *et al.*, 1995) versus natural speech-in-noise (0 dB SNR). Spectrally degraded speech simulates the primary information transmitted by cochlear implant devices. The results demonstrated that eight-band signals received benefit from context to the same extent as natural speech in noise. In contrast, four-band signals provided significantly less benefit from context than natural speech in noise. The most parsimonious explanation of this pattern of results is that listeners need a threshold amount of acoustic information to make use of context equally, regardless of the type of distortion. Alternatively, the results could be explained by assuming that listeners employ different strategies depending on the overall acoustic characteristics of the speech signal. Future experiments will distinguish between these alternatives.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday, 3 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Orlando
Acoustical Oceanography	7:30 p.m.	River Terrace 2
Animal Bioacoustics	7:30 p.m.	City Terrace 9
Architectural Acoustics	7:30 p.m.	Grand Ballroom 3
Physical Acoustics	7:30 p.m.	St. Johns
Psychological and Physiological Acoustics	7:30 p.m.	Grand Ballroom 7
Structural Acoustics and Vibration	7:30 p.m.	Daytona

Committees meeting on Wednesday, 4 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Clearwater
Signal Processing in Acoustics	8:00 p.m.	City Terrace 7

Committees meeting on Thursday, 5 November

Committee	Start Time	Room
Musical Acoustics	7:30 p.m.	Grand Ballroom 1
Noise	7:30 p.m.	Grand Ballroom 2
Underwater Acoustics	7:30 p.m.	River Terrace 2

Session 3aAAa

Architectural Acoustics and Musical Acoustics: Directivities of Musical Instruments and Their Effects in Performance Environments, Room Simulations, Acoustical Measurements, and Audio II

Timothy W. Leishman, Chair
Physics and Astronomy, Brigham Young University, N247 ESC, Provo, UT 84602

Invited Papers

8:00

3aAAa1. Directivities of musical instruments that relate to their effects in performance environments and room simulations. Larry Kirkegaard (Kirkegaard Assoc., 801West Adams St., Chicago, IL 60607, lkirkegaard@kirkegaard.com)

This paper addresses the particular aspects of Directivities of Musical Instruments that relate to "Their Effects in Performance Environments and Room Simulations." Large ensemble—small ensemble—soloist Flat floor—raked floor—risers—steep risers Wrap of the ensemble Relation to Soloists Amplified or acoustic Seated vs. Standing Effects for performers and effects for Audiences with various audience configurations. This paper is meant to be a celebration of new techniques combined with cautionary tales that speak to needs for "measuring equipment" that includes experienced ears and questioning minds engaged in designing the testing techniques and evaluating their results to extract the most pertinent and applicable information.

8:20

3aAAa2. Acoustical design of performance and rehearsal spaces influenced by instrument directivity. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

The sound that we hear in a music space is determined by the direct sound, as well as the early and late reflections from the room's surfaces, so instrument directivity and reflection control are of primary interest. Spatially directed sound by individual instruments and musical ensembles is controlled by the selection and placement of absorptive, reflective, and diffusive boundary surfaces, which contribute to the room's preference for both the musicians and the audience. We will describe how these surfaces are measured and characterized for directional and random incidence and how they have been used in many types of musical environments. In this presentation, we will discuss and present before and after measurements in the design of an individual practice room addressing room modes; describe the essential ingredients in an ensemble rehearsal space; present before and after perception questionnaires for a variable acoustics modular performance shell for small music ensembles and a symphony orchestra; describe a stage canopy optimization as a function of the support objective measure and present computer model coverage predictions addressing rear wall and overhead canopy treatments in an auditorium.

8:40

3aAAa3. Using multi-channel anechoic recordings to represent source directivity in room acoustics models to improve auralizations. Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu), Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE), and Jens Holger Rindel (Odeon A/S, Lyngby, Denmark)

A series of studies were conducted to investigate the effects of different sound source directivity representations on perceived differences in auralizations. The first study (Wang & Vigeant 2008) showed that subjects could not distinguish between auralizations created using both omnidirectional and measured static directivities of individual instruments. Differences may not have been observed since source directivities were not available for all octave bands of interest (125–8000 Hz). Significant differences were perceived between the auralizations generated using an omnidirectional source and a highly directional source for all bands. For the second study (Vigeant, Wang, & Rindel 2011), the effects of using 4- and 13-channel solo instrument anechoic recordings versus single channel recordings on realism and source width were evaluated. In general, the results showed an improvement in realism and decrease in source width with an increasing number of channels. In the final study (Vigeant, Wang, & Rindel 2008), 5-channel solo instrument anechoic recordings for two orchestral pieces were used to investigate the effects of using multi-channel and multi-source representations of an orchestra on realism, source depth, source width, and ease of distinguishing between instrument parts. For some of the cases studied, significant differences were found with the most complex representation of the orchestra.

9:00

3aAAa4. Quantification of time varying directivity of musical instruments in an orchestral context. Madeline A. Davidson and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska- Lincoln, 6349 Cedar Plaza #222, Omaha, NE 68106, madeline.davidson@huskers.unl.edu)

At the University of Nebraska, previous research has been done in the quantification of time varying directivity of played musical instruments using 13 channel anechoic recordings of solo passages. The proposed quantifiers involve measures of the extent to which directivity changes in magnitude as well as how often the directivity changes in spatial direction, using various time windows of analysis from 0.2 to 1 s. In this paper, these proposed quantifiers are applied to 5 channel anechoic recordings of a full orchestra for a Brahms symphony and a Mozart symphony. With silent sections removed from the recordings, the time varying directivity properties of musical instruments are more accurately examined within the context of played notes from two stylistically different symphonies.

9:20

3aAAa5. Interactive room auralization of sound sources with exchangeable directivities. Lukas Aspöck and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen, Kopernikusstr. 5, Aachen D-52074, Germany, las@akustik.rwth-aachen.de)

Although established measurement methods and simulation models for source directivities are available, there is a lack of tools to directly experience the audible effects of virtual sound sources in different conditions. Only an auralization in connection with psychoacoustic experiments can answer questions regarding the required solution and quality of sound source directivities. This paper presents a real-time simulation engine based on geometrical acoustics which accounts for both source directivities and the room acoustics of the environment. A hybrid simulation model is applied combining an image source model and a ray tracing algorithm to generate a binaural room impulse response, which is then convolved with an anechoic sound file (e.g., of a musical instrument). A convenient user interface is introduced by integrating the simulation engine into the 3D modeling program SketchUp. This makes it possible to easily position virtual sound sources and select various source directivities inside a modifiable room model. The listener receives immediate perceptual feedback and is able to examine the effects of varying virtual sound source locations and source directivities in different room acoustic environments.

Contributed Paper

9:40

3aAAa6. Recent advancements in massive multi-channel auralization. Jens Ahrens and Hagen Wierstorf (Univ. of Technol. Berlin, Ernst-Reuter-Platz 7/TEL 18, Berlin 10587, Germany, ahrjens@gmail.com)

Massive multi-channel auralization approaches like Wave Field Synthesis and Higher Order Ambisonics experienced a pronounced hype in the late 2000s during which the primary research goal was maximizing the physical accuracy of the synthetic sound fields that they create. The hype eventually faded as the achievable advancements turned out to be limited due to fundamental restrictions. Though, activities are still being pursued in the domain with the focus shifted towards perception of synthetic sound fields. This talk gives an overview over current activities, which aim at understanding localization, timbre, and spatial impression in general. The results show that localization performance in synthetic sound fields is close to the performance in real sound fields. Timbre and spatial impression exhibit impairments that are directly linked to the physical limitations of the employed systems. Promising options for improvements regarding the synthesis of artificial reverberation are discussed.

Session 3aAAb

Architectural Acoustics: Worship Space Acoustics: Three Decades of Design

David T. Bradley, Cochair
Physics + Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604

Erica E. Ryherd, Cochair
University of Nebraska, 1110 S 67th St., Architectural Engineering, Omaha, NE 68182-0816

Lauren M. Ronsse, Cochair
Audio Arts and Acoustics, Columbia College Chicago, 33 E. Congress Pkwy, Suite 601, Chicago, IL 60605

Chair's Introduction—9:35

Invited Papers

9:40

3aAAb1. The soundscape of worship. Gary W. Siebein and Kara A. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Soundscape theory provides insights into the architectural acoustic design of religious buildings that demonstrate how acoustical issues can be integrated within worship spaces of the 21st century. Eliade (1959) conceives of worship spaces as a sacred enclosure where communication with the God is made. The spaces must have a door to the world above, by which the God can descend to earth and man can symbolically ascend to heaven. Sometimes this is manifest as one confronts his or her God alone, in silence; sometimes, it is manifest in the joyous singing and praying or sad wailing of the entire community joined in corporate celebration or solemnity during momentous events. If one pays careful attention to the group involved with the design of a facility, they can tailor the architectural, acoustical, and electronic systems design to provide a unique acoustical identity that expresses the individual aspirations of the particular church. The technical tools for measurement and assessment of existing conditions, modeling of proposed changes, and simulating the anticipated results so evaluations can be made while the design is still in a computer are readily available to allow consultants to develop acoustical aspects of worship spaces that are unique for each project.

10:00

3aAAb2. How classical ideas inform 21st century architectural acoustics. Timothy Foulkes (Cavanaugh Tocci Assoc, Sudbury, MA) and John Prokos (Gund Partnership, 47 Thorndike St., Cambridge, MA 02141, JohnP@gundpartnership.com)

This presentation was inspired by John Prokos' essay in *Worship Space Acoustics: 3 Decades of Design*. It will be a joint presentation by Prokos, an architect and Foulkes, an acoustician. It was Marcus Vitruvius Pollio, the great First Century Roman architect and engineer, who first set forth the famous triumvirate of *firmness* (structure), *commodity* (usefulness), and *delight* (beauty) as the ideals of the architect. Even in today's digital age, these three tenets are as applicable as ever, nowhere more so than in architectural acoustics. This session will consider examples of buildings and spaces whose acoustics are as compelling as their visual design. For *firmness*, we will look at several examples including Gund Partnership's athletic center at Kenyon College, whose dramatic vaulted glass and steel structure nonetheless allow excellent acoustics. For *commodity*, among the projects we present is the Young Israel Symagogue in Brookline, MA, a collaboration between the two presenters, which has the flexibility to accommodate varied worship services and gatherings to meet the congregations's changing needs. Among the buildings demonstrating *delight* is Cavanaugh Tocci's work on the 1100 seat Oratory at Ave Maria University in Florida, a classically inspired church that seeks to replicate the acoustics of the great European cathedrals.

10:20

3aAAb3. Balancing sight and sound in heritage worship buildings. Dan Clayton (Clayton Acoust. Group, 2 Wykagyl Rd., Carmel, NY 10512-6224, mail@claytonacoustics.com)

Buildings for traditional religious gathering and time-honored worship practice require a balance of highly valued acoustical qualities such as reverberance for liturgical music, ensemble for choral singing, responsiveness for congregational participation, and clarity for intelligible speech. Geometry, dimensions, proportions, cubic volume, and boundary materials are critical elements of acoustical success. Many old European worship buildings are admired for their particularly fine blending of these criteria, and often cited as benchmarks for how buildings in the United States should look and sound. Although many heritage U.S. worship buildings look like their European precedents in terms of layout, shape, and size, a critical design element was modified, upsetting the acoustical balance. Boundary materials became lighter and thinner, making construction easier, faster, and less expensive. Acoustical upgrade of these existing buildings in the context of historic preservation/restoration brings additional design complexity, requiring equal measures of compromise from owner, users, acoustician, architect, and engineers. The acoustician's dilemma becomes "what we see may not be what we hear," as acquired acoustical expectations are upended by actual conditions. This paper will describe these differences and explore approaches to acoustical enhancement within limitations of the buildings themselves plus further constraints of contemporary preservation practice.

10:40

3aAAb4. Worship space acoustics: Design needs, trends, and issues for sacred spaces, relative to traditional and contemporary worship styles. Scott Riedel and Craig Schaefer (Acoust. Consultants, Scott R. Riedel & Assoc., Ltd., 819 N Cass St., Milwaukee, WI 53202, riedel@riedelassociates.com)

Concerns and goals for worship space acoustic design include the need for clear, intelligible speech, well blended, balanced, and projected music, and encouraging robust participation by the congregation in sung and spoken parts of a service. Also important are the attenuation of noise and control of acoustical anomalies that could interrupt, mask, or distort the critical expressions of speech and music. Achieving these goals includes careful attention to such factors as various cubic air volumes, unique geometric forms, functional proximities, and interior finish material selections. These factors are important to both improving existing rooms and in creating new designs. A current challenge is that of accommodating different musical styles (from traditional compositions and instruments to contemporary and ethnic forms) within the same space. Practical issues may include client understanding of differences between A/V systems and their capabilities vs. architectural room acoustics, and realities of budget limitations and value engineering. Before and after case-study examples exploring these goals, issues, and design solutions will be presented and discussed, including examples from the new publication, "Worship Space Acoustics, Three Decades of Design."

11:00

3aAAb5. Practical lessons in acoustical design learned from recent worship space projects. Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

In the last 25 years of worship space acoustics consulting projects (and building on experiences of the 25 years before that), consultants at Acentech have learned many important lessons about these unique and important spaces. Throughout that time period, the evolution of worship style has necessitated thoughtful new design approaches to worship space acoustics. Acentech consultants have learned lessons regarding the complex interplay between support for congregational singing and appropriate reverberation control for contemporary worship music. The equally complex interplay between traditional music and intelligibility of a preacher's speech has also taught profound and lasting lessons. Silence plays a critical and fundamental role in promoting a contemplative and prayerful environment—another important lesson learned. Using examples from the Acoustical Society's new Worship Spaces book, this presentation will illustrate practical lessons from these and other projects in Acentech's recent experience with worship facilities.

11:20

3aAAb6. Worship space acoustics: Three decades of design. Walid Tikriti (Acoustonica, LLC, 33 Pond Ave., Ste. 201, Brookline, MA 02445, wtikriti@acoustonica.com)

The presentation discusses the acoustics and noise control design for new construction and existing worship space renovation and the challenges during the design stage. The first project is an addition of a chapel to an existing church. Treating the hard surfaces on the walls and ceiling proved challenging due to the historic nature of the building. The second project is an addition of a 1800 sq. ft. multi-purpose room with a full glass wall. The Glory Be Hall space is intended to provide an acoustically pleasant space for various programs. The third project is a new construction of a 550 seat worship space. The acoustics design focused on achieving a balance between the amplified sound system, the background noise level from the mechanical equipment, the noise isolation from adjacent spaces including nursery and public spaces.

11:40

3aAAb7. Design and renovation of worship spaces at Fourth Presbyterian Church in Chicago. Dawn Schuette and Jennifer Nelson Smid (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, jnelson@thresholdacoustics.com)

Fourth Presbyterian Church is among Chicago's most iconic buildings. Designed by Ralph Adams Cram, it was dedicated in 1914. Threshold Acoustics worked alongside Gensler on a 5-story addition to the Fourth Church facilities that provided needed worship and program space in celebration of their centennial. The Genevieve and Wayne Gratz Center was included as a case study for the publication "Worship Spaces Acoustics: Three Decades of Design." It houses the 350-seat Buchanan Chapel, which is used as overflow for holiday services, intimate worship services, Children's Chapel, music performances, and weddings. Threshold Acoustics is currently working with Fourth Church on the restoration of their historic Skinner Organ in the sanctuary. An acoustic study of the sanctuary was completed to evaluate and determine the current challenges the organ faces to carry sound to all locations within the Nave. The renovations to the organ and chamber are underway and are to be finished by December 2015 for holiday services. This paper will describe Threshold's acoustic design for the Gratz Center and Sanctuary.

Session 3aAO

Acoustical Oceanography: Munk Award Lecture

Andone C. Lavery, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

Chair's Introduction—10:55

Invited Paper

11:00

3aAO1. Ocean acoustic tomography: Past, present, and maybe future. Carl Wunsch (Earth and Planetary Sci., Harvard Univ., Cambridge, MA 02138, carl.wunsch@gmail.com)

This talk will describe some of the origins of what became known as ocean acoustic tomography, and how it has evolved in the intervening years. Some published mis-conceptions about its beginnings are somewhat interesting. The possibilities of its having a more central role in the determining the evolution of the ocean in future climate raise fascinating technical problems that probably have equally fascinating solutions.

Session 3aBAa

Biomedical Acoustics: Sonothrombolysis

Kevin J. Haworth, Cochair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

Kenneth B. Bader, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586

Chair's Introduction—8:00

Invited Papers

8:05

3aBAa1. Non-invasive thrombolysis using histotripsy beyond the "intrinsic" threshold (microtripsy). Xi Zhang (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Rm. 1107 Gerstacker Bldg., Ann Arbor, MI 48109), Hitinder Gurm (Interventional Cardiology, Univ. of Michigan, Ann Arbor, MI), Gabe Owens (Pediatric Cardiology, Univ. of Michigan, Ann Arbor, MI), and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI 48109, zhenx@umich.edu)

Histotripsy has been demonstrated as a non-invasive, drug-free, image-guided thrombolysis method using cavitation alone. Microtripsy is a new histotripsy approach, where cavitation is generated using 1-cycle ultrasound pulses with negative pressure exceeding a threshold intrinsic to the medium. We hypothesize that, using microtripsy, cavitation can be generated entirely within the vessel lumen, without contacting the vessel wall. Microtripsy was used to create a flow channel through a clot in a vessel-mimicking phantom by scanning the therapy focus through the clot at a pre-set interval. Different scan intervals, doses, and strategies (single focus vs. electrical-steered multi-foci) were tested in both unretracted and retracted clots. The flow channel size, thrombolysis rate, and clot debris particle sizes were measured. A cavitation cloud and flow channel was successfully generated and completely confined within the 6.5 mm-diameter vessel lumen through unretracted and retracted clots. A 1-4 mm channel was created in unretracted clots, at thrombolysis rate of 3.3 min/cm. Using an electrical-steered multi-foci method, a 1–2.5 mm channel was generated through retracted clots, at thrombolysis rate of 5.5 min/cm. The debris particles generated were no greater than 150 μm. The results show the potential of microtripsy for precise and effective clot recanalization, minimizing risks of vessel damage and embolism.

8:25

3aBAa2. Insights into mechanisms of sonothrombolysis using high speed imaging. Flordeliza Villanueva (HVI, UPMC, 200 Lothrop St. PUH A351, Pittsburgh, PA 15213, villanuevafs@upmc.edu)

Thrombotic arterial occlusion is the major cause of acute cardiovascular syndromes such as stroke and myocardial infarction. Exposing the thrombus to microbubbles in the presence of ultrasound facilitates thrombus disruption, and is thus a potentially powerful therapeutic strategy for thromboembolic diseases. However, optimization of "sonothrombolysis" is constrained by a lack of understanding of underlying mechanisms for blood clot disruption in response to ultrasound-induced microbubble vibrations. We tested the hypothesis that inertial cavitation induces mechanical clot disruption by optically characterizing lipid microbubble interactions with thrombus + ultrasound using an ultra-high speed microscopy imaging system capable of imaging at MHz frame rates. A microscope/acoustic stage was used to hold an experimentally created thrombus and microbubbles, which were insonified (1 MHz) during synchronized high speed imaging. Large amplitude microbubble oscillations in response to an inertial cavitation regime caused thrombus deformation and pitting. These data implicate a direct mechanical effect of oscillating microbubbles on mediating clot disruption.

8:45

3aBAa3. Combined lysis of thrombus with ultrasound and systemic tissue plasminogen activator for emergent revascularization in acute ischemic stroke (CLOTBUST-ER): An update. Andrei V. Alexandrov (Neurology, UTHSC, 855 Monroe Ave., Ste. 415, Memphis, TN 38613, avalexandrov@att.net)

Background: Continuous exposure of intracranial arterial occlusions to pulsed wave ultrasound enhances tissue plasminogen induced recanalization. Our hypothesis is that sonothrombolysis can improve functional outcomes of stroke patients receiving tPA therapy. Methods: The primary objective is to evaluate the efficacy of a novel transcranial ultrasound device and systemic tPA (Target) compared to systemic tPA alone (Controls) in subjects with acute ischemic stroke and NIHSS scores 10 or greater. This is a randomized (1:1), placebo-controlled, multi-site, phase 3 clinical trial to evaluate the efficacy and safety of a novel ultrasound device, as an adjunctive therapy to tPA treatment in subjects with acute ischemic stroke: total projected enrollment 824 patients with interim analyses at 1/3 and 2/3 of enrollment. Current Status: CLOTBUST-ER had active enrollment at approximately 70 sites in 14 countries worldwide. DSMB recommended to stop the trial after second interim analysis. Functional outcomes are still being assessed via modified Rankin Scores at 90 days and the primary endpoint will be analyzed using ordinal shift statistical analysis. Final results will be presented at ISC 2016. Conclusions: CLOTBUST-ER is the first phase three multinational randomized blinded clinical trial of sonothrombolysis for treatment of acute ischemic stroke. ClinicalTrials.gov Trial Registry ID: NCT01098981.

9:05

3aBAa4. Microbubble pumps: Ultrasound theragnostic agents. Christy K. Holland, Himanshu Shekhar, and Kenneth B. Bader (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng. Program, Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Cardiovascular disease is the number one cause of death worldwide, and thrombo-occlusive disease is a leading cause of morbidity and mortality. Ultrasound has been developed as a tool to induce the release, delivery, and enhanced efficacy of a thrombolytic drug (rt-PA) and bioactive gases from echogenic liposomes. By encapsulating drugs into micron-sized and nano-sized liposomes, the therapeutic can be shielded from degradation within the vasculature until delivery is triggered by ultrasound exposure. Insonification accelerates clot breakdown in combination with rt-PA and ultrasound contrast agents, which nucleate sustained bubble activity, or cavitation. Mechanisms for ultrasound enhancement of thrombolysis, with a special emphasis on cavitation and radiation force, will be reviewed. The delivery of bioactive gases from echogenic liposomes to promote vasodilation and cytoprotection will also be discussed.

9:25

3aBAa5. Microscale interactions between ultrasound stimulated microbubbles and the fibrin networks of clots. David Goertz, Christopher Acconcia (Univ. of Toronto, S665a, 2075 Bayview Ave., Toronto, ON m4k1x5, Canada, goertz@sri.utoronto.ca), and Ben Leung (Sunnybrook Res. Inst., Toronto, ON, Canada)

While the concept of microbubble mediated sonothrombolysis is now well established, a detailed mechanistic understanding of this process remains both elusive as well as necessary in order to facilitate the development of improved exposure methods. As lytic effects may ultimately arise from microscale bubble-clot interactions, we have employed high-speed microscopy and two-photon microscopy to examine these interactions first in (transparent, fluorescently tagged) fibrin clots, and then in blood clots. Bubble "population" studies in fibrin clots show the prominent role of primary and secondary radiation forces: bubbles are first directed toward the clot boundary, where their concentrations increase and interactions such as clustering and coalescence occur frequently. A subset of bubbles penetrate into the clots, disrupting the fibrin network structure along their paths. Once initiated, the resulting tunnels act in subsequent exposures as conduits for bubbles to enter and access deeper points within the clot. Using optical tweezers, individual bubble experiments reveal that bubble entry into the clots, along with the accompanying network damage and fluid uptake, are a function of the network pore size, bubble size, and the exposure scheme. With blood clots, the erosion surface evolves in a complex manner, involving the ejection of erythrocytes and the development and progression of a cell depleted fibrin network zone. The characteristics of the erosion front are highly dependent on exposure conditions.

9:45

3aBAa6. Molecular mechanisms of the effect of ultrasound on the fibrinolysis of clots. Irina N. Chernysh (Dept. of Cell & Developmental Biology, Univ. of Pennsylvania, 1154 BRB, 421 Curie Blvd., Philadelphia, PA 19104-6058), E C. Everbach (Eng. Dept., Swarthmore College, Swarthmore, PA), Prashant K. Purohit (Dept. of Mech. Eng. and Appl. Mech., Univ. of Pennsylvania, Philadelphia, PA), and John W. Weisel (Dept. of Cell & Developmental Biology, Univ. of Pennsylvania, Philadelphia, PA 19104-6058, weisel@mail.med.upenn.edu)

Our objective was to identify mechanisms for the ultrasound-enhanced acceleration of tissue-type plasminogen activator (t-PA)-induced fibrinolysis of clots. Measurements of turbidity as a function of time, used to characterize quantitatively the effects of ultrasound, showed that the ultrasound pulse-repetition frequency affected clot lysis times, but there were no thermal effects. Ultrasound in the absence of t-PA produced a slight but consistent decrease in turbidity, suggesting a decrease in fibrin diameter due solely to the action of ultrasound, likely caused by an increase in protofibril tension because of vibration from ultrasound. Changes in fibrin network structure during lysis with ultrasound were visualized in real time by deconvolution microscopy, revealing that the network becomes unstable when 30–40% of the network was digested, whereas without ultrasound, the fibrin network was digested gradually and retained structural integrity. Fluorescence recovery after photobleaching during lysis was used to characterize the kinetics of binding/unbinding and transport. Ultrasound causes a decrease in the diameter of fibers due to tension as a result of vibration, leading to increased binding sites for plasmin(ogen)/t-PA. The positive feedback of this structural change together with increased mixing/transport of t-PA/plasmin(-ogen) is likely to account for the observed enhancement of fibrinolysis by ultrasound.

10:05–10:20 Break

Contributed Papers

10:20

3aBAa7. Histotripsy for liquefaction of large extravascular hematomas for fine-needle aspiration: Feasibility study. Tatiana D. Khokhlova, Thomas Matula, Yasser Haider, and Wayne Monsky (Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98104, tdk7@uw.edu)

Intra- and extra- muscular hematomas result from repetitive injury, or sharp and blunt limb trauma. There are currently no short-term treatment options for large hematomas, only lengthy conservative treatment. The goal of this work was to evaluate the feasibility of histotripsy to liquefy extravascular hematomas for subsequent fine-needle aspiration. Cavitation histotripsy (CH) and boiling histotripsy (BH) were applied to liquefy 50-mL clots in an *in vitro* gel phantom. 1 MHz and 1.5 MHz transducers were used with duty cycles of 1–2% for BH and 0.3–0.5% for CH, pulse durations 10 ms for BH, and 3.3–5 çs for CH, treatment duration 5–60 s and peak focal pressures of $p^+ = 70–100$ *MPa*, and $p^- = 15–25$ *MPa*. The liquefied lysate was aspirated with a 18–21 gauge needle. The lysate was analyzed by histology and sized in a Coulter counter. Under the same exposure duration, BH lesions were 1.5–2 times larger than CH lesions. 99% of lysate particulates were smaller than 10 çm. CH-aided liquefaction was slower, but the voids were more regularly shaped, facilitating easier aspiration. A combination of BH and CH may be most optimal for liquefaction. [Work supported by NIH K01 EB 015745 and Washington State Life Science Discovery Fund (Grant No. 3292512).]

10:35

3aBAa8. Fibrin-targeted echogenic liposomes for localized ablation of thrombi with histotripsy pulses. Kenneth B. Bader, Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586, Kenneth.Bader@uc.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Tao Peng, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX), and Christy K. Holland (Internal Medicine, Univ. of Cincinnti, Cincinnati, OH)

Deep vein thrombosis is a debilitating condition that can result in pulmonary embolism, or post-thrombotic syndrome. Clot lysis can be initiated with histotripsy, a novel form of therapeutic ultrasound that uses the mechanical action of microbubble clouds to ablate tissue. Thrombolytic-loaded echogenic liposomes (t-ELIP) have fibrin targeting capabilities and also entrain gas microbubbles that act as cavitation nuclei. We hypothesize that t-ELIP can nucleate microbubble cloud formation near the clot, providing targeted ablation. Highly retracted porcine whole blood clots were exposed to either histotripsy pulses alone, or histotripsy pulses and t-ELIP between peak negative pressures of 7.5 MPa and 20 MPa. Thrombolytic efficacy was assessed via clot mass before and after treatment. Microbubble cloud activity was monitored with passive cavitation imaging (PCI) during histotripsy exposure. For a given pressure, no significant differences were observed in the thrombolytic efficacy with or without t-ELIP. The dimensions of the bubble cloud, assessed on the PCIs, were significantly reduced in the presence of t-ELIP ($p < 0.05$). Thus, the presence of t-ELIP confines the business end of the bubble activity to the clot. Overall, the combination of histotripsy and t-ELIP is a promising mechanism for targeted ablation of thrombi resistant to thrombolytic drugs.

10:50

3aBAa9. Ultrasound-enhanced thrombolysis: Mechanistic observations. Marie de Saint Victor, Dario Carugo, Constantin Coussios, and Eleanor P. Stride (Univ. of Oxford, IBME, Old Rd. Campus, Oxford, United Kingdom, marie.desaintvictor@pmb.ox.ac.uk)

Ultrasound and microbubbles have been widely demonstrated to accelerate the breakdown of blood clots, but the mechanisms of treatment require further investigation. In particular, there is a need to clarify the effect on the fibrin matrix—the insoluble polymer mesh that determines a clot's integrity and mechanical properties. The objective of this *in vitro* study was to observe in real-time the mechanisms of microbubble-enhanced sonothrombolysis at the microscale. Fluorescently labeled porcine plasma clots were prepared on a glass coverslip and exposed to different types of microbubbles with or without the fibrinolytic agent recombinant tissue plasminogen activator. A 1 mm thick piezoelectric element was coupled with the glass substrate and driven at the resonant frequency of the system (1.9 MHz), with a duty cycle of 5% and a 0.1 Hz pulse repetition frequency. The acoustic field within the clot was characterized using a fiber optic hydrophone. Changes in the fiber network were monitored for 30 min by confocal microscopy.

Session 3aBAb

Biomedical Acoustics: Therapeutic Ultrasound, Microbubbles, and Bioeffects I

Michael Bailey, Chair
Center for Industrial and Medical Ultrasound, Applied Physics Lab, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

11:15

3aBAb1. High throughput acoustic and optical characterization of microbubbles for optimized contrast ultrasound imaging. Paul Rademeyer and Eleanor Stride (IBME, Oxford Univ., Old Rd. Campus, IBME, Oxford OX3 7DQ, United Kingdom, paul.rademeyer@eng.ox.ac.uk)

Echogenic particles, such as microbubbles and volatile liquid micro-/nanodroplets, have shown considerable potential in a variety of clinical diagnostic and therapeutic applications. The accurate prediction of their response to ultrasound excitation is however extremely challenging, and this has hindered the optimization of techniques such as quantitative ultrasound imaging and targeted drug delivery. Existing characterization techniques, such as ultra-high speed microscopy, provide important insights, but suffer from a number of limitations; most significantly difficulty in obtaining large data sets suitable for statistical analysis and the need to physically constrain the particles, thereby altering their dynamics. Here, a microfluidic system is presented that overcomes these challenges to enable the measurement of single echogenic particle response to ultrasound excitation with a throughput of 20 samples/second and an uncertainty below 7% in the measurements. Demonstration of optimized contrast ultrasound imaging is shown based on the characterization of over 100,000 individual SonoVue® bubbles.

11:30

3aBAb2. Design, characterization, and performance of a dual aperture, focused ultrasound system for microbubble-mediated, non-thermal ablation in rat brain. Jonathan Sutton, Yui Power, Yongzhi Zhang, Natalia Vykhodtseva, and Nathan McDannold (Dept. of Radiology, Brigham and Women's Hospital, 221 Longwood Ave., Boston, MA 02115, jtsutt@bwh.harvard.edu)

Submegahertz ultrasound (US) with microbubbles (μB) can ablate brain tissue: a low amplitude (< 1 MPa) alternative to thermal ablation. Single element transducers at transcranial frequencies have broad axial profiles compared to the size of targets in small animals. Thus, we sought a system to ablate millimeter volumes in normal and tumorous rat brain non-thermally using US and μB. The system consisted of two transducers oriented at 120°, driven at different frequencies (F = 837 kHz, ΔF = 30 Hz) to reduce the depth-of-field by 78%. To monitor cavitation, a passive detector (650 kHz) was confocally aligned with the therapy field. Targets were registered stereotactically. μB injections (100, 200, and 400 μl/kg) with 5-minute sonications proceeded at acoustic pressures relative to the *in vivo*cavitation threshold (0.3–0.6 MPa) determined *a priori*. Following MRI and sacrifice (1 h, 4 days, 10 days), tissue was fixed and stained. At 1 h, small lesions (<2 mm) were selectively comprised of stenotic capillaries with macrophage infiltration and neuronal damage. As lesions grew, stenotic capillaries within this escalating margin increased, along with neuronal damage. T2* lesions larger than 4 mm exhibited necrosis, cyst, and tissue removal histologically after four days. Persistent, strong (>5 dB) inharmonic energy indicated formation of large ablation volumes along the US path. Low level (<2 dB) or sporadic (<50%) cavitation indicated incomplete ablation. Results and mechanisms, especially with respect to ischemic stroke, will be discussed.

11:45

3aBAb3. An empirical model of size-isolated ultrasound-triggered phase shift emulsions. Karla P. Mercado, Lindsay Snider, Kirthi Radhakrishnan, and Kevin J. Haworth (Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu)

High-speed mechanical agitation is commonly used to produce microbubbles and droplets for ultrasound imaging and therapy. This technique results in a high concentration (~10^{10} particles/mL) of polydisperse particles (less than 400 nm to greater than 15 μm in diameter). Differential centrifugation has been used to isolate microbubbles and droplets of specific sizes. In our prior work, we have isolated droplets between 2 μm and 5 μm. In the current work, we have isolated different sizes of droplets by adjusting centrifugation speeds. Our size-isolation protocol increased the fraction of droplets between 1 μm and 3 μm from 8% for non-centrifuged droplets to 87% for differentially centrifuged droplets. An empirical model for the size distribution after differential centrifugation was developed. The measured fraction of droplets in the supernatant and pellet for all sizes after a single centrifugation was used in the empirical model. There was a 3% difference in the volume-weighted mean diameter of the experimentally measured and empirically modeled size distributions. The coefficient of variations of the experimentally measured and empirically modeled size distributions were 24% and 22%, respectively. The empirical model allows for determining appropriate centrifugation parameters to obtain desired size distributions. [Work supported in part by NIH grant KL2 TR000078.]

Session 3aEA

Engineering Acoustics: Test Facilities and Acoustic Calibration

Roger M. Logan, Chair
Teledyne, 12338 Westella, Houston, TX 77077

Chair's Introduction—8:00

Invited Papers

8:05

3aEA1. ANSI/ASA S1.20-2012, "Procedures for Calibration of Underwater Electroacoustic Transducers," Revision Details. Robert Drake (NUWC, P.O. Box 5029, Newport, RI 02841, orl_preds@yahoo.com)

The ANSI/ASA S1.20-2012 standard, "Procedures for Calibration of Underwater Electroacoustic Transducers," was released in February 2012. It is a revision of American National Standard S1.20-1988 (R2003). An overview of the content of this standard inclusive of the typical primary and secondary open water procedures for determining the measurable characteristics of free-field receive voltage sensitivity and transmitting responses are examined along with highlights of materials that are new to the 2012 revision. In this later category, this presentation provides overview information contained in the S1.20-2012 related to correction factor application (with an emphasis on extension cable usage) and introductory material on measurement uncertainty analysis (along with the identification of common error sources). A cursory look at the informational annexes will also be addressed with an emphasis on the 2012 revision-specific ones related to medium correction factors, nonlinear effects including cavitation, and a standard-target method for calibrating active sonars.

8:25

3aEA2. Underwater acoustic calibration facilities at Applied Research Laboratories, The University of Texas at Austin. Richard D. Lenhart (Appl. Res. Labs., The Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, lenhart@arlut.utexas.edu) and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The calibration of various underwater sources and receivers is routinely undertaken at The University of Texas at Austin's Applied Research Laboratories using an open water facility, the Lake Travis Test Station, and both indoor and outdoor test tanks on the ARL main campus. Calibrations are also conducted using standing wave facilities and transducer electrical input impedance measurements can be obtained. These facilities and procedures will be described.

8:45

3aEA3. Standard-target calibration of sonars for imaging and scattering measurement: An ISO-standard development. Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu) and Christian de Moustier (10dBx LLC, San Diego, CA)

Given a standard target of known acoustic scattering properties, the combined transmit–receive transfer characteristics of an active sonar system can be determined by measuring the response due to the target placed at specified positions in the sonar transmit and receive beams. These measurements are repeatable to ± 0.1 dB, establishing an absolute reference when measuring the scattering properties of unknown targets. This standard-target sonar-calibration method is in worldwide use with scientific echo sounders and multibeam sonars that provide the water-column signal, yet no international metrological standard defines it. To prescribe protocols for this method, which has a demonstrated applicability over the frequency range from 1 kHz to several megahertz, an effort was initiated in 2014 within the International Organization for Standardization (ISO) Technical Committee 43 (Acoustics) Subcommittee 3 (Underwater Acoustics). The associated Working Group 4 (Standard-target method of calibrating active sonars) has formulated an outline of the intended ISO standard. The elements of this will be reviewed.

9:05

3aEA4. Acoustic calibration in mass production. Roger M. Logan (Teledyne, 12338 Westella, Houston, TX 77077, rogermlogan@sbcglobal.net)

While calibrations in high volume settings are bound by the same laws of nature that govern laboratory grade measurements, the laws of economics bring their own set of challenges to the bench. This presentation will explore the art and science of making efficient acoustic measurements in small air-filled tanks using mostly COTS equipment.

9:25

3aEA5. Investigation and correction for the calibration error of using two-microphone impedance tube method on low absorption materials. Yi Zhang, Tongan Wang, and Kristopher Lynch (Acoust., Vib. & Community Noise, Gulfstream Aerosp., 500 Gulfstream Rd., Savannah, GA 31408, Yi.Zhang1@gulfstream.com)

The two-microphone impedance tube method has been widely used for determining the sound absorption coefficients of materials. In this method, a calibration procedure is performed to correct the amplitude and phase mismatch of the two microphones, through repeated measurement of a specimen with the two channels interchanged. Both ASTM E1050-12 and ISO 10534-2 suggest the use of a highly absorptive material as the calibration specimen and the calibration, once complete, is valid for all successive measurements. When following the Standards, however, test materials of low absorption capability exhibited a pattern of peaks and dips in the absorption coefficient curve at certain frequencies, which are obviously not a part of the material's own behavior. This paper will discuss the cause of these peaks and dips and also suggest a simple solution to the issue. Using the new method, one is able to obtain smooth absorption coefficient curves for low absorption test materials just as expected.

9:40

3aEA6. Calibration of high frequency MEMS microphones and pressure sensors in the range 10 kHz–1 MHz. Sébastien Ollivier (LMFA - UMR CNR 5509, Univ. Lyon 1, Ctr. Acoustique - Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully 69134, France, sebastien.ollivier@univ-lyon1.fr), Petr V. Yuldashev (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Cyril Desjouy (LMFA - UMR CNRS 5509, Ecole Centrale de Lyon, Ecully, France), Maria Karzova, Edouard Salze (LMFA - UMR CNRS 5509, Ecole Centrale de Lyon, Lyon, France), Alexandra Koumela, Libor Rufer (TIMA Lab. (CNRS, G-INP, UJF), Grenoble, France), and Philippe Blanc-Benon (LMFA - UMR CNRS 5509, Ecole Centrale de Lyon, Ecully, France)

In the context of both nonlinear acoustics, and downscaled acoustic or aero-acoustic experiments, the characterization of the high frequency response of microphones and pressure sensors remains a critical challenge. In the case of the design of new MEMS microphones and shock pressure sensors with response in the frequency range of 10 kHz–1 MHz, this question was addressed by the definition of a new calibration method based on a spark source that generates spherical weak shock acoustic pulse. Waves are short duration non-symmetric N-waves with duration of about 40 microseconds and front shock rise time of the order of 0.1 microsecond. Taking advantage of recent works on the characterization of such pressure waves using an optical interferometer, and considering non linear propagation of weak shockwaves, we were able to estimate the incident pressure wave in the range of 10 kHz–1MHz. Hence, from the output voltage of the microphones, the frequency response was obtained in this range. The method applies whatever the transduction principle and the sensor mounting. [Work supported by the French National Agency for Research (SIMI 9, ANR 2010 BLANC 0905 03, and LabEx CeLyA ANR-10-LABX-60/ANR-11-IDEX-0007).]

9:55–10:10 Break

10:10–10:30 Panel Discussion

Session 3aED

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

Preston S. Wilson, Cochair
Mech. Eng., The University of Texas at Austin, 1 University Station, C2200, Austin, TX 78712

Joseph F. Vignola, Cochair
Mechanical Engineering, The Catholic University of America, 620 Michigan Ave., NE, Washington, DC 20064

Posters will be on display and all authors will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aED1. Experimental investigation of nonlinear wave behavior in a tensegrity mast. Joy Westland, Andrew A. Piacsek, and Peter Zencak (Dept. of Phys., Central Washington Univ., Ellensburg, WA 98926-7422, westlandj@cwu.edu)

First described by Fuller and Snelson in the mid-twentieth century, tensegrity structures comprise a network of elements that alternately support compression (struts) and tension (cables). When the cable elements are prestressed, tensegrity structures maintain a stable form, regardless of orientation. If external forces are applied, the structure will deform both locally and globally, with the stress distributed throughout the structure. Removal of the external forces will result in the structure regaining its original shape. However, the deformation is not a result of elastic strains in the individual elements, but rather a reorientation of the elements, and the restoring force is nonlinear. To characterize the effect of this nonlinearity on dynamic behavior, a tensegrity mast was constructed consisting of 60 aluminum bars, each of length 11 cm; the total mast length is approximately one meter. A mechanical shaker mounted at one end is used to drive the mast into resonance, as well as to create longitudinal pulses. Frequency response measurements are consistent with a stiff nonlinearity and evidence is shown for harmonic generation during pulse propagation. The effect of pre-stress tension is also investigated and discussed.

3aED2. Electroacoustical evaluation of a prototype headphone amplifier for educational purposes. Lúcia C. da Silva (Centro de Tecnologia, Universidade Federal de Santa Maria, Rua Lauro Bulcão, 1715, São Sepé, RS 97340-000, Brazil, lucodasil@hotmail.com.br), Ricardo Brum, Moisés dos Santos Canabarro (Centro de Tecnologia, Universidade Federal de Santa Maria, Santa Maria, RS, Brazil), and Stephan Paul (Centro de Engenharia da Mobilidade, Universidade Federal de Santa Catarina, Florianópolis, SC, Brazil)

In several classroom activities, it is required to feed a signal simultaneously to several headsets to have students listening to the same sound signal, e.g., to show binaural effects. To drive several headphones, an appropriate headphone amplifier is needed. The aim of this work was to build a prototype of a simple headphone amplifier based on a publicly available project around different types of operational amplifiers and test its electrical and acoustical performance when used with typical headphones. Frequency response function, total harmonic distortion, and signal-to-noise ratio were evaluated. Among the different OPAMPs the OPA2134 showed the best performance, especially because of the very flat frequency response function in the hearing range, low harmonic distortion, and superior SNR (−83 dBFS). Acoustical test were therefore performed based on the amplifier with the OPA2134 only, using a Sennheiser HD 600 headphone and a Sennheiser KR4 microphone. THD measured with this configuration ranged from −86 dBC@2.5 kHz and global SPL = 70 dB to −35 dBC@100 Hz and global SPL = 110 dB. At all, the test showed that the amplifier circuit is appropriate for the required application, taking into account that it is very simple and can be mounted easily by the students.

3aED3. Acoustic and sediment data in the southern New England Bight. Chitanya Gopu (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, chitanya@bu.edu), Gopu R. Potty, James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and James F. Lynch (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Wood Hole, MA)

This study provides a review of the acoustic and ocean bottom sediment data collected during the Shelf Break Primer experiment conducted in 1996. The location of the experiment was in the southern New England Continental Shelf called the "New England Mud Patch." The mud patch is a 13,000 square kilometer area covered by fine-grained sediment. Previous surveys in this area have estimated the thickness of this fine grained sediment deposit to be as much as 13 m. This layer of sediment rests on a reflector that is geomorphically similar to and continuous with the Holocene transgressive sand sheet, which is exposed on the shelf to the west of this area. This fine-grained sediment layer, which is oriented in an east-west direction seaward of the 55–65 m isobath, contains more than 30% silt and clay. During the Primer experiment, broadband acoustic sources were deployed in the western side of the "mud patch" along and across the continental shelf. The acoustic data collected on a vertical line array will be analyzed. Sediment data from gravity cores from the area will also be presented. [Work sponsored by Office of Naval Research, code 322 OA.]

3aED4. Estimation of the effective perceived noise level during approach of Brazilian AMX A-1 military aircraft. Bernardo H. Murta, Gil F. Greco (Universidade Federal de Santa Maria, Rua Dezenove de Novembro, 289, 302, Santa Maria, Rio Grande do Sul 97060160, Brazil, be.murta@gmail.com), Stephan Paul (Universidade Federal de Santa Catarina, Florianopolis, Brazil), Jean C. Bernardi, and Matheus Pereira (Universidade Federal de Santa Maria, Santa Maria, Brazil)

Aircraft noise is an important cause of annoyance mainly in regions close to airports or military airbases. Civil aircraft data of sound emission, such as EPNL, are commonly available to the public; however, for military aircrafts such data are not easily accessible. Noise emission is commonly quantified in terms of Equivalent Perceived Noise Level during approach, low altitude overflight, and simulated take-off. This study presents the results of the estimation of the Equivalent Perceived Noise Level (EPNL) for the AMX-A1 aircraft during approach at Santa Maria's Airbase, in south Brazil. Since there is an University and its Hospital on the surroundings of the airbase, the approach is the most annoying situation. The measurements were made at the vicinities of the airbase based on the equivalent procedures established by ICAO where calibrated wave signals were recorded and processed, leading to results of EPNL of the order of 134 EPNdB. The calculated levels were found to be coherent for military aircrafts and shall be validated with further investigation.

3aED5. Sonothrombolysis of porcine blood clots using 1 MHz pulsed ultrasound. Atousa Nourmahnad, Luke Barbano, and Erich C. Everbach (Eng. Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, eceverbach@comcast.net)

Blood clots block proper blood flow in vessels and often lead to fatal health conditions such as cardiac ischemia and stroke, especially in an aging population. Because ultrasonic fields can cause cavitation of fluids, administration of ultrasound and microbubble contrast agents can induce intravascular thrombolysis (dissolution of blood clots). Quantitative and qualitative data were collected on how a porcine blood clot dissolves when exposed to Definity™ microbubbles and 1 MHz pulsed ultrasound. In addition, bubble dynamics in response to the measured ultrasonic wave field were modeled using a modified Gilmore equation. The results allowed us to optimize parameters and better understand the interaction of the clot fibrin structure and the movement of microbubbles through it. Advances in the understanding of sonothrombolysis can help transform its clinical application in patients efficiently, especially in instances where life depends on rapid dissolution of a thrombus.

WEDNESDAY MORNING, 4 NOVEMBER 2015 GRAND BALLROOM 1, 8:30 A.M. TO 11:20 A.M.

Session 3aNS

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: Role of Fit-Testing Systems for Hearing Protection Devices

Melissa Theis, Cochair
Air Force Research Laboratory, 426 Crusader Drive, Dayton, OH 45449

Elizabeth McKenna, Cochair
Air Force Research Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Chair's Introduction—8:30

Invited Papers

8:35

3aNS1. Operational challenges associated with the use of fit test systems in the U.S. Air Force. Melissa Theis (Oak Ridge Inst. for Sci. and Education, 426 Crusader Dr., Dayton, OH 45449, melissa.theis.1.ctr@us.af.mil) and Elizabeth McKenna (The Henry M. Jackson Foundation for the Advancement of Military Medicine, Wright Patterson AFB, OH)

Hearing protection fit test systems provide immediate and direct feedback on the goodness-of-fit for a specific individual based on the personal attenuation rating (PAR) obtained from the measurement. Fit test systems may provide improved retention of hearing protection training, particularly when used in combination with face-to-face training. One challenge for the U.S. Air Force is determining when face-to-face training should occur, and how detailed this training should be to maximize the benefits for end users. A second challenge is determining if various hearing loss patterns render PAR data inaccurate, because many Air Force personnel have hearing loss. The ability of individuals with hearing loss to achieve a desired PAR for a workplace may be influenced by the dynamic range of the system, particularly when measuring the protected condition. This presentation will discuss how the two challenges mentioned above must be investigated before implementing fit test systems as a hearing preservation tool for U.S. Air Force personnel, and discuss some preliminary data obtained in these areas.

8:55

3aNS2. Assessing attitudes toward use of hearing protection devices and effects of an intervention based on results of fit testing. Pegeen Smith (Tech. Service, 3M Co., 620 Holly Rd., Cadillac, MI 49601, psmith4@mmm.com)

As field attenuation estimation systems (FAESs) have become more prevalent, there has been a call from industry to quantify the value of including them into hearing conservation programs. This presentation describes a study that assessed attitudes toward the use of hearing protection devices (HPDs) and the effect of an educational intervention based on results of fit testing by comparing the personal attenuation rating (PAR50) before and after intervention. Employees (n = 327) from a large metal container manufacturer at four geographic locations were tested with a FAES to identify workers (n = 91) needing the intervention. PAR50 values significantly increased from baseline to post-intervention ($p < 0.001$, 15.1 to 26.9) and at six-month follow-up ($p < 0.001$, 95% CI = −11.2, −6.3). However, perceived self-efficacy (SE) scores in using HPDs significantly declined from baseline to post-intervention ($p = 0.006$, 95% CI = 0.3, 1.9) but were not significantly related to PAR50. Therefore, a FAES can assist the health and safety professional to identify workers who are at high risk (low PAR50), teach the proper fit and use of HPDs, and enhance hearing protector selection.

9:15

3aNS3. Viability of large-scale fit-testing in the U.S. Navy and Marine Corps. Jeremy Federman (SUBMED, NSMRL, Bldg 141 Trout Ave, NAVSUBASE NLON, Groton, CT 06349-5900, jeremy.s.federman.civ@mail.mil)

In support of the Navy and Marine Corps initiatives to prevent military and civilian personnel from suffering hearing loss due to noise exposure, and to ensure hearing readiness and fitness for duty in the military and civilian workforce, this project explored the potential to implement a fit testing procedure for objectively measuring hearing protection devices (HPDs) performance on individuals. Due to the relative uniformity of noise exposure during recruit training, relevant training and experience, and likelihood of participants having hearing within normal limits, 320 training recruits at USMC MCRD Parris Island were included in Phase 1 and 900 in Phase 2. Factors of interest were fit testing pass-fail rates, test duration, and training. The pass-fail criterion attenuation level was set at 25 dB based on the output level of the M4 weapon used during basic training (Amrein & Letowski, 2012; NIOSH, 2014). Both test duration and PAR estimation accuracy were considered in evaluating the practicality of implementing a fit-testing program. Results of objective fit testing were considered as a metric of training effectiveness and HPD functionality.

9:35

3aNS4. Hearing protection fit-test system pilot study: Results and recommendations for the hearing conservation program. Quintin Hecht and Matthew Williams (Public Health Consultation, USAF School of Aerosp. Medicine, 2510 5th St., Wright-Patterson AFB, OH 45433-7913, quintin.hecht.1@us.af.mil)

Since the inception of hearing protection device fit-test systems, long-term training benefits have been scarcely investigated. A hearing protection fit-test system was evaluated for performance and training benefits in a pilot study among U.S. Air Force members randomly sampled from those in the hearing conservation program. Subjects were randomly assigned to control or intervention. Both groups received an initial and a 6-month follow-up appointment. At the initial appointment, the control group received standard training for hearing protection, while the intervention group was fit-tested, trained, and fit-tested again. At the follow-up appointment, both groups received fit-testing. Results of the intervention group's initial appointment will be examined and the two groups' follow-up fit-test results will be compared. These findings will be discussed in relation to effects on training and retention. Methods for conducting this study and a descriptive analysis of the subject population will also be reviewed. Recommendations for fit-testing integration into a hearing conservation program will be provided based on our study results and other literature.

9:55–10:10 Break

10:10

3aNS5. Comparison sound-field measurements of hearing protector attenuation and fit-test systems. William J. Murphy and David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

The American National Standards Institute (ANSI), accredited S12 standards committee for noise, working group 11 is developing a standard to characterize the performance of hearing protector fit test systems. An inter-laboratory study was conducted to compare sound field measurements of real-ear attenuation at threshold (REAT) tests of hearing protection with three hearing protection fit-test systems. The draft ANSI standard for personal attenuation ratings from fit-test systems includes adjustments to the fit test system personal attenuation rating for the accuracy and precision of the measurement. As well, adjustments are proposed for the effect of fitting and for spectral variability of effective attenuation. In the inter-laboratory study, two fit-test systems exhibited precision of about 2 dB while the third system had a precision of about 4 dB. The third system underestimated the sound field attenuation by about 2 dB. Because the inter-laboratory study tested two fittings of the hearing protector with each fit-test system, the variability of individual fitting is described as an average of the differences of first and second tests. This paper will also evaluate the effect of the noise spectrum for the three fit-test systems.

10:30

3aNS6. Earplug fit testing: Practical examples. Theresa Y. Schulz (Honeywell Safety, 9218 Brookwater Circle, College Station, TX 77845, Theresa.Schulz@Honeywell.com)

Preventing noise-induced hearing loss at the workplace appears deceptively easy. Well-intentioned professionals often assume that providing hearing protectors and administering audiograms are, by themselves, preventive measures. But occupational hearing loss often continues unabated. This presentation shows data from companies that have used best practice techniques to stop noise-induced hearing loss with fit-testing and individual training as their hearing conservation tools. These tools can change the attitudes and behaviors of earplug users, young and old.

10:50–11:20 Panel Discussion

Session 3aPA

Physical Acoustics: General Topics in Physical Acoustics I

Michael R. Haberman, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Contributed Papers

8:00

3aPA1. Transmission and reception of ultrasound via a polymer film. Hironori Tohmyoh and Shota Mukaimine (Dept. of Nanomechanics, Tohoku Univ., Aoba 6-6-01, Aramaki, Aoba-ku, Sendai, Miyagi 980-8579, Japan, tohmyoh@ism.mech.tohoku.ac.jp)

We report on the changes in the waveform of the echo which was transmitted and received via a polymer film inserted between the water and the solid samples. Generally, the frequency characteristics of an ultrasonic transducer are fixed because the piezoelectric element embedded in the transducer is unchangeable. This study is aimed to enhance the flexibility of ultrasonic transducers by freely changing the frequency components of the echo. For this purpose, we paid attention to the frequency dependence of the echo transmittance among three media. A theoretical model for the ultrasonic transmission system comprising the water, the film and the solid sample was developed and the validity of the model was verified by experiments where the ultrasound excited by the ultrasonic transducer was transmitted into the steel samples via a polymer film. The amplitude of the echo was enhanced by the insertion of the film and its frequency components were modulated toward the higher frequency side. It was experimentally confirmed that the waveform of the echo obtained from the sample via a film could be accurately predicted by the theoretical model developed.

8:15

3aPA2. Uncertainty calculations of pressure sensitivity of one inch microphones using pressure field method at the National Metrology Institute of Egypt. Rabab S. Ahmed, Hany A. Shawky, Rabab S. Youssef, Tarek M. El-Basheer, and Hatem K. Mohamed (Acoust., National Inst. for Standards (NIS), Tersa St., El Harm, Giza, Egypt, Cairo 12211, Egypt, ruby01986@yahoo.com)

Primary calibration method is used by relatively few laboratories such as national calibration laboratories and few large automotive, space, or governmental organizations, which work at high technological level. The National Metrology Institute of Egypt (NIS) has developed a pressure calibration system that used for calibration of pressure sensitivities. In this study, we estimate the uncertainty of unknown one inch condenser microphone using two known references one inch condenser microphones according to international standard. IEC 61094-2009 gives more information's and details on the uncertainty calculations. In this method, a wide frequency range with a high accuracy and repeatability were achieved.

8:30

3aPA3. The research on piezoelectric cylindrical oscillating transducer for dipole acoustic logging. Yinqiu Zhou, Xiuming Wang, and Yuyu Dai (Inst. of Acoust., Chinese Acad. of Sci., 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhouyinqiu@mail.ioa.ac.cn)

Acoustic transducer, which is a core component of acoustic logging tool, determines the development of acoustic logging technology. In order to realize the goal of dipole acoustic logging transducer with the features of low frequency, broadband and high radiation power, a cylindrical oscillating dipole acoustic transducer using an actuator of a group of different trilaminar piezoelectric bending bars is designed and studied. Modal and harmonic response analyses of the whole transducer are conducted by using the finite element method, including analyses of input conductance of transducer in air and water, transmitting voltage response and directivity patterns. Besides, the proposed transducer is fabricated and tested to compare with numerical simulation results. Moreover, the numerical results of the proposed transducer are compared with that of traditional dipole transducer. The comparison result indicates the proposed dipole transducer is characterized by low frequency, broadband, and high radiation power, which is feasible in the application of dipole acoustic logging.

8:45

3aPA4. Manipulation and levitation of particles with acoustic vortices. ZhenYu Hong (Dept. of Appl. Phys., Northwestern PolyTech. Univ., Xi'an, China), Asier Marzo, Sri Subramanian (Dept. of Comput. Sci., Univ. of Bristol, Bristol, United Kingdom), and Bruce W. Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, University Walk, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

Acoustic waves with screw dislocations at their wavefronts, or acoustic vortices, are characterized by an azimuthal phase dependence. The time average of an acoustic vortex forms a circular region of high pressure which can be used to trap particles, and by manipulating the vortex axis, it can controllably translate them. In addition, acoustic vortices carry orbital angular momentum which can be transferred to absorbing particles causing them to rotate. This paper describes various new observations of these phenomena. In the first example, microparticles are manipulated in a host liquid at high ultrasonic frequencies (2 MHz), and a controlled translation and rotation is demonstrated. In this viscously dominated system, small particles rotate slowly with the liquid and larger particles are drawn into the centre of the vortex. The particle dynamics are explained as a fine balance between the transfer of angular momentum and the action of acoustic radiation forces. In the second example, much larger particles are levitated in air in the low ultrasonic range (40 kHz) and controlled translation and rotation is again observed. Here, viscosity is shown to play a lesser role in the particle dynamics, and conditions are observed under which the particles are rapidly ejected from the vortex core.

9:00

3aPA5. Induced radiation force and torque on a viscoelastic particle in an ideal fluid. José P. Leao and Glauber T. Silva (Phys., Federal Univ. of Alagoas, Av. Lourival Melo Mota, sn, Maceio, Alagoas 57035-557, Brazil, glauber@pq.cnpq.br)

The interaction of an acoustic wave with a suspended particle may produce radiation force and torque on the particle through linear and angular momentum transfer. Theoretical analysis of the radiation force exerted on a rigid, compressible fluid, elastic solid, and layered spherical particle is abundantly found in the literature. Nevertheless, less attention has been devoted to cases involving a homogeneous viscoelastic particle, even though viscoelastic materials are ubiquitous. Here, the radiation force and torque exerted on this kind of particle is studied in detail using the fractional Kelvin-Voigt viscoelastic model. Analytical expressions are obtained in the monopole-

dipole approximation to a small particle in the long-wave limit. Considering a traveling plane wave interacting with a high-density polyethylene (HDPE) particle, we found that the axial radiation force is negative, i.e., the force and the wave propagation are in opposite directions. For a first-order Bessel vortex beam, a full 3D tractor beam develops on the HDPE particle placed in the beam's axis. Furthermore, negative axial radiation torque also appears on the particle, i.e., the torque and the beam's angular momentum are contrariwise. In addition, possible implications of this method to the advancement of acoustophoretic techniques will be outlined.

9:15

3aPA6. A modal analysis view of minimum phase response of acoustic enclosures. Sahar Hashemgeloogerdi and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., Rochester, NY 14627, shashemg@UR.Rochester.edu)

When the response of an acoustic system is minimum phase, the existence of a stable and causal inverse is assured, which enables compensation for acoustic effects such as reverberation. In this paper, we explore the ranges of minimum phase behavior of one, two, and three-dimensional acoustic enclosures in light of their modal frequency response. We show that a one-dimensional acoustic enclosure, which by definition contains only axial modes, is always minimum phase. The responses of two and three-dimensional acoustic enclosures are characterized by a combination of normal modes in multiple dimensions. We show that two and three-dimensional acoustic enclosures demonstrate minimum phase behavior over limited frequency intervals. Minimum phase response tends to be confined to low frequencies, and a cutoff frequency below which the enclosure is minimum phase is identified.

9:30

3aPA7. A computationally efficient method for the frequency-domain analysis of visco-thermal acoustic propagation in arbitrary geometries. Gustavo Martins and Julio A. Cordioli (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Campus Universitário, Trindade, Florianópolis, SC 88040-970, Brazil, julio.cordioli@ufsc.br)

The traditional approach used to represent the visco-thermal acoustic propagation within arbitrary geometries is the numerical solution of the full linearized Navier-Stokes (FLNS) equations. The FLNS model, usually solved by means of the Finite Element method (FEM), requires a large number of DOFs to be considered at each node. Therefore, the FLNS becomes much more computationally expensive than the standard acoustic model. An alternative is the sequential linear Navier-Stokes (SLNS) model, which can reduce the computational cost of the solution. However, the SLNS can be still computationally expensive for some types of analysis such as optimization's procedures. The SLNS model is obtained by means of some simplifications and decoupling of the FLNS equations and requires the solution of a viscous and a thermal dimensionless scalar field. The computational cost of the SLNS approach is manly due to the requirement of the solution of these scalar fields at each discrete frequency. In this paper, a semi-analytical solution of these parameters is proposed and tested in acoustic problems with arbitrary geometries. The new approach shows precision similar to the SLNS model, while the computational cost is only slightly higher than the standard acoustic model.

9:45

3aPA8. The scattering of sound by a buried obstacle in an extended reaction ground. Yiming Wang and Kai Ming Li (Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, wang1679@purdue.edu)

The boundary integral equation (BIE) method is widely used in outdoor sound scattering problems due to its computational efficiency when compared with the finite element method. The advantage of the BIE is apparent for the case when the scattering surface is much smaller than the computational domain of the problem. However, one of the major difficulties of the BIE formulation is the need to obtain the required Green functions for a sound source placed in the vicinity of the extended surface—both above and below the air/ground interface. There have been some recent developments where accurate and fast computations of these Green functions become available for use in the BIE formulation. This paper discusses some of these accurate asymptotic solutions and addresses the required steps to develop the BIE formulation. The computation of sound fields for the obstacle buried partially/totally in an extended reaction ground are presented.

10:00–10:15 Break

10:15

3aPA9. Turbulence effects on broadband pulses propagating near the ground. D. Keith Wilson, Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Sandra L. Collier, Jericho E. Cain (U.S. Army Res. Lab., Adelphi, MD), and Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint Louis, France)

Broadband sound pulses (such as from gunfire and explosions) distort randomly as they propagate through atmospheric turbulence. Recently, a general theory was formulated to describe the statistical moments, including the mean and space-time coherence, of such signals propagating along line-of-sight paths (Ostashev *et al*., J. Acoust. Soc. Am. **136**(5), 2414–2431 (2014)). This presentation focuses on some practical issues related to the physical interpretation of the theoretical predictions for the moments, such as what the theory predicts for pulse signals and turbulence regimes of typical interest, and how the behavior of individual pulse events is reflected in the averages that can be compared directly to the theory. In particular, we examine the conditions when the scattering is dominated by pulse *wander* (variations in arrival time) and when is it dominated by pulse *spread* (broadening of the energy in the impulse). Conversion of real-valued signals to analytic signals, and the averaging of these representations, is also discussed.

10:30

3aPA10. Wind turbine radiated acoustic signals—Propagation in a temporally and spatial variable marine meteorological boundary layer. Marshall H. Orr (College of Earth, Ocean and Environment, The Univ. of Delaware, PO Box 254, Bryantown, MD 20375, rubyspiral@gmail.com), Kenneth E. Gilbert, Xiao Di (National Ctr. for Physical Acoust., The Univ. of MS, University, MS), and Mohsen Badiey (College of Earth, Ocean and Environment, The Univ. of Delaware, Newark, DE)

The temporal and spatial properties of an acoustic field radiated from a wind turbine operating in a marine meteorological boundary layer and a spatial temporal variable impedance boundary will be quantified using paraxial equation based numerical simulations. A sound speed field will be constructed using wind field measurements made with a 915 MHZ radar wind speed sensor and local temperature and humidity measurements obtained from a meteorological tower. Temporally variable impedance boundary conditions for a tidal marine marsh will be included in the simulations.

10:45

3aPA11. Infrasonic wind noise reduction comparison for wind fences and porous domes. JohnPaul Abbott (NCPA and Dept. of Phys. and Astronomy, Univ. of MS, 122 PR 3049, Oxford, MS 38655, jrabbott@go.olemiss.edu), Richard Raspet (NCPA and Dept. of Phys. and Astronomy, Univ. of MS, University, MS), John Noble, W. C. K. Alberts, and Sandra Colliler (U.S. Army Res. Labs., Adelphi, MD)

This paper reports on an investigation directly comparing the measured wind noise and detected acoustic signals for two types of co-located infrasonic wind noise reduction barriers. The first type is a set of cylindrically shaped wind fence enclosures and the second type is a set of 2.0 m diameter semi-porous fabric domes. The fence configurations included variations to the height, width, and number of filtering layers. Forty and fifty-five percent porosities were used for each of these configurations. Wind Noise reductions for the domes and the 5 m diameter fence are comparable at low wavenumbers, while the reductions for the 5 m fence is better at high wavenumbers. The porous domes and the 6 m high wind fence achieve comparable maximum reduction levels; however, the 6 m high fence mitigates noise for a broader wavenumber range. The 10 m diameter fence achieves the best reduction levels overall, while the porous domes are better at mid-range wavenumbers. Additional filtering layers improve all reductions and show the greatest improvement for the fences and the higher porosity dome.

11:00

3aPA12. Influence of source motion, wind, and temperature profiles on the effective impedance of an absorptive surface. Kai Ming Li and Bao N. Tong (Mech. Eng., Purdue Univ., 140 South Russell St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

This paper presents a theoretical study of the sound field due to a moving source placed above an absorptive surface in a stratified medium in the presence of wind and sound speed profiles. The current analysis starts with the general equations for an isentropic inviscid flow field. A monopole source is assumed to be traveling parallel to the absorptive surface at a constant speed. The standard Lorentz transform can be applied and the Fourier decomposition can be used to express the sound field in an integral form that is amenable to further mathematical treatments. Either the fast field computation or the method of steepest descent can be used to evaluate the integral. A two-dimensional formulation was considered initially, but it can subsequently be extended to three-dimensional flow fields. The effect of source motion, wind, and temperature profiles on the acoustic properties of the absorptive surfaces have been explored.

11:15

3aPA13. Statistical moments of impulse propagation through the atmosphere. Jericho E. Cain, Sandra L. Collier (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jericho.cain@gmail.com), Sylvain Cheinet (French-German Res. Inst. of Saint-Louis, Saint-Louis, France), Vladimir Ostashev, and D. K. Wilson (US Army Engineer Res. and Development Ctr., Vicksburg, MS)

In order to localize and classify propagating impulsive signals, a theoretical model that incorporates effects due to atmospheric turbulence, ground impedance, wind speed, wind direction, and refraction is needed to guide the analysis of data measured with acoustic sensing systems. Recently, measurements with planar arrays were made that form a database of acoustic measurements of impulse sources propagated through various atmospheric conditions, ranges, and ground types. This paper presents analyses of the statistical moments of this data for the purposes of comparison with recent theoretical models.

11:30

3aPA14. Empty cavity in a cavitating liquid: Features of flow structure. Valeriy Kedrinskiy (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru) and Ekaterina S. Bolshakova (Phys., Novosibirsk State Univ., Novosibirsk, Russian Federation)

Problem of empty cavity dynamics in a two-phase medium is considered. The initial equilibrium state of "cavity-medium" system is disturbed when pressure inside cavity falls abruptly up to 0. Rarefaction wave arising on an interface initiates a cavitation development. Two-phase mathematical model is applied to investigate the medium state dynamics. The medium parameters correspond to a distilled water state: microbubbles, 1.5 μm, their density 10^6 cm^{-3}, and gas concentration, about 10^{-5}. The numerical analysis has shown that interface "cavity-medium" becomes a cavitating spherical layer. The concluding process of cavity collapse can be characterized by two stages. First, the interface as a spherical layer in a result of its cumulation is transformed into a spherical bubbly cluster with 1 mm radius. Cluster contains $2.5\cdot10^5$ cм$^{-3}$ microbubbles with 40 μm radii. Gas concentration is distributed from 20%,cluster center, up to 1% on its surface. Second, the cumulation of flow on the spherical bubbly cluster will determine a level of internal energy of compressed bubbly cluster and its further dynamics. The similar phenomenon was found in the experiments on the development, structure, and collapse of a rupture forming in cavitating layer of distilled water at its shock-wave loading. The analysis of experimental data has shown that a rupture in the cavitating layer is the cavity with interface as thin layer of cavitating liquid and its collapse tends to the bubbly cluster formation. [Support RFBR, grant 15-05-03336.]

11:45

3aPA15. Acoustic measurements of the noise generated by the Silver Fox Unmanned Aerial System. Frank S. Mobley (Human Effectiveness Directorate, U.S. Air Force, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil)

Acoustic measurements of the noise generated by the Silver Fox Unmanned Aerial System (UAS) were accomplished on a test fixture at Owens-Corning. These measurements, made in one-percent throttle increments, revealed a region over which the noise power curve was linear. Source noise directivity patterns were constructed for each throttle increment using a spherical harmonic series expansion and compared to directivity patterns constructed using a proposed linear interpolation methodology. Moreover, these predictions from the two source construction methods were compared to validation measurements and demonstrate that the interpolation method is viable for spherical harmonic source representations.

Session 3aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Potpourri (Poster Session)

Frederick J. Gallun, Chair
National Center for Rehabilitative Auditory Research, VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aPP1. Hearing protection device field attenuation estimation systems. JR W. Stefanson (Auditory Protection and Performance Div., US Army Aeromedical Res. Laboratory/ Hearing Ctr. of Excellence, 6901 Farrel Rd., Fort Rucker, AL 36362, earl.w.stefanson.ctr@mail.mil) and William A. Ahroon (Auditory Protection and Performance Div., U.S. Army Aeromedical Res. Lab., Fort Rucker, AL)

The United States Army Aeromedical Research Laboratory has conducted comparative, developmental, and evaluation studies for Hearing Protection Device Field Attenuation Estimation Systems (HPD FAES). The Army's interest in protecting Soldiers' hearing led to the development of the Communications Earplug (CEP), which couples communication systems with a foam earplug in an attempt to block hazardous noise while still passing to the user the clearest speech signal attainable. A recent developmental study addresses the need to fit-test the CEP when used with foam or custom earplugs in a quick and objective fashion using a field-microphone in real-ear (F-MIRE) procedure. Commercial systems have also been evaluated along with a clinical audiometer used by the Army Hearing Program for application as an HPD FAES. A select few FAESs were studied for their attenuation measurement accuracy as compared to the laboratory standard ANSI/ASA S12.6. Completed and ongoing studies will be presented on the FAESs' utility, cost, and effectiveness.

3aPP2. Characteristics of 40,000 calls to the National Hearing Test. Charles S. Watson (Res., Commun. Disord. Technol., Inc., CDT, Inc., 3100 John Hinkle Pl, Bloomington, IN 47408, watson@indiana.edu), Gary R. Kidd (Speech and Hearing, Indiana Univ., Bloomington, IN), Jill E. Preminger (Dept. of Otolaryngol. Head and Neck Surgery and Commun. Disord., Univ. of Louisville, Louisville, KY), James D. Miller, Daniel P. Maki, and Alex Crowley (Res., Commun. Disord. Technol., Inc., Bloomington, IN)

The National Hearing Test (NHT) is a telephone-administered screen for hearing loss (Watson *et al*., J. Am. Acad. Audiol., 2012) that obtains thresholds for three-digit sequences in a noise background. The NHT has been validated by comparing threshold SNR values to mean pure-tone loss. During five weeks in 2014 the NHT was offered without charge to the general public. Over 40,000 calls were made after articles describing the test appeared in eight large-circulation and 19 smaller circulation newspapers, estimated to reach less than 10% of the US public. Among those who completed the test, 81% were estimated to have clinically significant loss in one or both ears. Call numbers suggested that 88% were made from landlines and the remainder from cell phones. Threshold SNRs for cell phones were about 1.0 dB higher than those for landline phones. Samples of the callers were later contacted through telephone and email surveys. Responses indicated a positive influence in terms of the likelihood of seeking further evaluation and obtaining hearing aids if advised to do so. [Research supported by the National Institute for Deafness and other Communication Disorders of the National Institutes of Health under award number 5R44DC009719.]

3aPP3. Objective evaluation of the acoustic properties of various types of chest pieces in the modern acoustic stethoscopes. Karolina M. Nowak (Dept. of Endocrinology, Ctr. of Postgraduate Medical Education, Ceglowska 80 St., Warsaw 01-809, Poland, karolina.nowak@ippt.pan.pl) and Lukasz J. Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., Warsaw, Poland)

The acoustic properties of a stethoscope are largely determined by the construction of its chest piece, which can either be open or closed with a diaphragm. Different solutions are offered on the market and advertised for their advantages in sound quality. However, no objective data, neither supporting nor disproving the benefits resulting from implementing the specific features, are available. The aim of the present study is to provide such data. A laboratory stand for measuring velocities of vibrations of different points of diaphragms or skin surface during the actual auscultation examination was developed and constructed. Those vibrations are the primary source of the acoustic signal in a stethoscope, and thus, the obtained results provide important conclusions regarding the reasonability of using various types of terminals in chest pieces. It is shown, that thin, stiff diaphragms mounted on susceptible suspension rings ensure significantly better acoustic performance than the other investigated solutions. The influence of the shape and construction of the chest piece on the acoustic properties were evaluated during independent, complementary research, conducted using precision microphone placed in the earpiece of a stethoscope. The obtained results deny the common opinions regarding advantages of the bell-type chest pieces in low-frequency acoustic band.

3aPP4. On material properties and damping models for the dynamic modeling of the human middle ear by means of the Finite Element Method. Felipe S. Pires, Diego C. Arellano (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Campus Universitário, Trindade, Florianópolis, SC 88040-970, Brazil, felipesmp.emc@gmail.com), Stephan Paul (Mobility Eng. Dept., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil), and Julio A. Cordioli (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil)

An accurate dynamic model of the human middle ear is a valuable tool to better understand the mechanisms involved in the human hearing and some middle ear pathologies. It is also essential to the design and evaluation of implantable hearing devices, which may be connected to the middle ear structures. A considerable number of studies describe the development and validation of such models, and a review of these studies shows that there is a significant spread of material and dynamic properties used in the models. In this work, a detailed Finite Element model of the human middle ear, including tympanic membrane, complete ossicular chain, joints, and soft tissues (ligaments and tendons), is constructed and used to assess the influence of the input properties in the middle ear transfer functions. A frequency-dependent acoustic impedance at the oval window is used to represent the

cochlea. Ossicle motion predicted by the model is compared with experimental data from the literature, and the effects of different material properties and damping models are investigated. Although variations in material properties can have an impact in the middle ear transfer functions, different damping models shown to have a much larger effect, possibly leading to considerable errors in the numerical predictions.

3aPP5. Sound-induced flash illusion revisited. Aya C. Kito, Takafumi Furuyama, Kohta I. Kobayasi, Shizuko Hiryu, and Hiroshi Riquimaroux (Graduate school of life and medical Sci., Doshisha, 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 6100394, Japan, dmp1011@mail4.doshisha.ac.jp)

Auditory-visual illusion occurs when visual information and auditory information integrates in the brain. This is conceived because auditory information has temporal superiority than visual information. Purpose of the study is to investigate mutuality influence of auditory information and visual information. Subjects answered the number of sounds and flashes when the auditory and visual stimuli are presented. We selected five auditory and visual stimuli (percentage answered as 2 times were 0%, 25%, 50%, 75%, and 100%) from the auditory only and visual only case and combined the stimuli. Subjects answered the number of sounds and flashes when the stimuli are presented simultaneously. When comparing the rate of unilateral and bilateral case, auditory only case and a combination of ambiguous auditory stimuli and definite visual stimuli, auditory information tends to shift toward visual information. This result suggests that in this case, visual information has temporal superiority so auditory information can perceive slight difference because it is influenced by definite visual information and flash-induced sound illusion occurs. From this result, we can suggest that temporal illusion occurs from temporal ambiguity of stimuli rather than temporal resolution of modality.

3aPP6. Effects of aging on auditory duration discrimination. Rachael Luckett and Edward L. Goshorn (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, Hattiesburg, MS 39406-0001, rachael.luckett@eagles.usm.edu)

The effects of aging on auditory processing are well documented. The involved anatomy includes the entire auditory pathway from peripheral to central mechanisms. Existing procedures to identify auditory processing disorders include a wide variety of signals with distortions or reduced redundancy in the temporal or frequency domains or the addition of noise. This project investigated the effects of aging on the ability to discriminate duration of a musical auditory signal. A musical signal was used to add temporally-variant acoustical information that would not be present in a pure tone. A digitized.wav piano note (f_o = 422 Hz) with gradually decaying amplitude was edited in Sound Forge to produce two signals equal in spectrum, RMS amplitude, and VU level but differing durations (1000 and 800 ms). One hundred pairs of these musical signals were arranged in random order with half equal and half unequal in duration. The subject's task was to designate same or different. Subjects ranged in age from 18 to 90 years. A regressions analysis revealed that age is a significant ($p = .002$) coefficient for predicting performance on a duration discrimination task. The findings suggest that duration discrimination tasks may contribute to identification of auditory processing disorders in adults.

3aPP7. Statistical modeling of expected rates of permanent hearing loss in newborn infants with data derived from the center for disease control. Edward L. Goshorn, Charles G. Marx, Kimberly Ward, and Marietta Paterson (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, PsychoAcoust. Res. Lab., Hattiesburg, MS 39401, edward.goshorn@usm.edu)

The Center for Disease Control (CDC) began gathering data on Universal Newborn Hearing Screening (UNHS) in 1999 and had data from all states and territories by 2005. The purpose of UNHS is to identify permanent hearing loss (PHL) in infants and make referrals for intervention. Health-care professionals may use CDC data to monitor the effectiveness of screening/diagnostic programs. Effective monitoring may designate whether or not an "expected" number of infants has been identified for a given birth rate. However, because the incidence of PHL in newborn infants is so low (nine-year CDC average across all states and territories equals 0.0013), a relatively large number of infants may be screened before the first occurrence of PHL. Therefore, there is legitimate concern that true positives will be missed due to a high rate of true negatives. Clinically useful models for comparing observed to expected data are needed. CDC data from 2005 to 2013 were used to set boundaries for binomial and negative-binomial distributions. These derived distributions were used to produce tables/graphs showing expected values and confidence intervals for an appropriate range of incidences and quantity of infants screened. The authors also offer actions to pursue if observed data vary significantly from models.

3aPP8. Phonetically balanced and psychometrically equivalent monosyllabic word lists for word recognition testing in Thai. Sajeerat Poonyaban, Pasinee Aungsakulchai, Charturong Tantibundhit (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand), Chutamanee Onsuwan (Dept. of Linguist, Faculty of Liberal Arts, Thammasat Univ., Dept. of Linguist, Faculty of Liberal Arts, Thammasat University, Rangsit Campus, Khlong Luang 12120, Thailand, consuwan@hotmail.com), Rattinan Tiravanitchakul (Dept. of Commun. Sci. and Disord., Faculty of Medicine, Mahidol Univ., Ratchathewi, Bangkok, Thailand), Krit Kosawat (National Electronics and Comput. Technol. Ctr., Khlong Luang, Pathumthani, Thailand), and Adirek Munthuli (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Khlong Luang, Pathumthani, Thailand)

In speech audiometry, a common and crucial method to obtain a supra-threshold (dB) at which words are repeated with maximum accuracy is referred to as word/speech recognition testing. For Thai, Thammasat University and Ramathibodi Hospital Phonetically Balanced Word Lists 2015 (TU-RAMA PB'15) were created with five lists, each with 25 monosyllabic words. Besides its phoneme distributions being based on large-scale Thai spoken corpora [1], TU-RAMA PB'15 is in line with TU PB'14 [2], [3] with emphasis on phonetic balance, symmetrical phoneme occurrence, and word familiarity. To evaluate its homogeneity in terms of decibel intelligibility, the lists were recorded and presented to 10 normal hearing participants, ranging from 0 to 50 dB HL in 2 dB increments (ascending order) until they repeated correct verbal responses. Using logistic regression, regression slopes and intercepts were calculated to estimate percentage of correct performance at any given intensity and to construct psychometric functions for every list. Derived psychometric function slopes ranged from 0.2015 to 0.2262 while intensities required for 50% intelligibility ranged from 17.0876 to 20.8856. Two-way Chi-Square analysis performed on both parameters indicated that there was no significant difference among the five lists. Further testing is needed to verify the use among the hearing-impaired individuals.

3aPP9. Prevalence of acoustic reflexes in the United States. Gregory A. Flamme, Kristy K. Deiters, Stephen M. Tasko (Speech Pathol. and Audiol., Western Michigan Univ., 1903 W. Michigan Ave., MS 5355, Kalamazoo, MI 49008, greg.flamme@wmich.edu), and William A. Ahroon (Auditory Protection and Performance Div., US Army Aeromedical Res. Lab., Fort Rucker, AL)

The acoustic reflex is a contraction of the middle ear muscles in response to high-level sounds. Acoustic reflexes are invoked as a protective mechanism in some damage-risk criteria (DRC). However, acoustic reflexes are not always observed among people without auditory dysfunction, and should not be included in DRC unless there is 95% certainty that 95% of the population have acoustic reflexes. In the current study, we present the prevalence of acoustic reflexes among people 12 years and older ($N > 11,400$), using data from the National Health and Nutrition Examination Survey (NHANES). The NHANES can be used to produce prevalence estimates generalizable to the non-institutionalized U.S. population. Ipsilateral reflexes were screened at two elicitor frequencies and detected using Frequentist methods and via Kalman filtering of the reflex trace. Reflexes are pervasive only among those with hearing thresholds better than 15 dB HL at all frequencies, and fall below the criterion certainty with poorer sensitivity even at lower frequencies. Age and tympanometric variables are also related to reflex detection. Reflex prevalence is generally high among young people

with adequate hearing sensitivity for unrestricted military duty, but the prevalence is not uniform among audiometric configurations within this hearing profile.

3aPP10. Acoustic models of co-varying vocal roughness and breathiness. Mark D. Skowronski (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD 3004, Tampa, FL 32620, skowronski@usf.edu), Lisa M. Kopf (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Rahul Shrivastav (Office of the Vice President for Instruction, Univ. of Georgia, Athens, GA), and David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Dysphonia is characterized by several vocal qualities including breathiness and roughness. These qualities can coexist within a single voice. To examine the potential interaction between judgments of one voice quality in the presences of another, a large set of synthetic voices that co-varied along the breathiness and roughness continua were created and evaluated by 10 listeners. The perception of roughness (synthesized via amplitude modulation) was unaffected by degree of breathiness. The perception of breathiness, however, was affected by the degree of roughness: for low-breathy voices breathiness increased by about 2 dB as roughness increased from no roughness to maximum roughness; yet for high-breathy voices, breathiness decreased by about 2 dB as roughness increased. Current acoustic models of voice quality (pitch strength, auto-correlation peak, cepstral peak, partial loudness ratio, and glottal-noise-excitation ratio) do not explain the observed interaction of roughness and breathiness. To explain the observed interaction, we evaluate the possibility that roughness perception is dominated by temporal cues (which are unaffected by the presence of breathiness) and that breathiness perception is driven by spectro-temporal cues which are affected by the sub-harmonics generated by amplitude modulation. The sub-harmonics partially contribute to non-harmonic energy (increasing breathiness) and partially mask aperiodic energy (decreasing breathiness).

3aPP11. The effects of acoustic variability on absolute pitch categorization: Evidence of contextual tuning. Stephen C. Van Hedger, Shannon L. Heald, and Howard C. Nusbaum (Psych., The Univ. of Chicago, 5848 S. University Ave., Beecher 406, Chicago, IL 60637, stephen.c.hedger@gmail.com)

Absolute pitch (AP) is defined as the ability to label or produce a musical note without the aid of a reference note. Despite the large amounts of acoustic variability encountered in music, AP listeners generally experience perceptual constancy for different exemplars within note categories (e.g., recognizing that a C played on a tuba belongs to the same category as a C played on a piccolo). The present studies investigate whether AP possessors are sensitive to context variability along acoustic dimensions that are not inherently linked to the typical definition of a note category. In a speeded target recognition task, AP participants heard a sequence of notes and pressed a button whenever they heard a designated target note. Within a trial the sequence of notes was either blocked according to note-irrelevant variation or contained a mix of different instruments (Experiment 1), amplitude levels (Experiment 2), or octaves (Experiment 3). Compared to the blocked trials, participants were significantly slower to respond in the mixed-instrument and mixed-octave trials, but not the mixed-amplitude trials. Importantly, this performance difference could not be solely attributed to initial performance differences between instruments, amplitudes, or octaves. These results suggest that AP note identification is contextually sensitive.

3aPP12. Motor suppression of the auditory system extends to the brainstem frequency following response and is mediated by attentional demands. Serena Klos and Howard C. Nusbaum (Psych., The Univ. of Chicago, 5848 S University Ave., Chicago, IL 60637, sklos@uchicago.edu)

Neural theories of auditory perception often characterize subcortical structures as relay stations by which acoustic input is passively encoded into a representation that can be recognized by cortical networks. However, efferent projections throughout the peripheral auditory pathway (Huffman & Henson, 1990) suggest a corticofugal network consisting of ascending and descending pathways may play an important role in the perception of acoustic signals. Based on cortical suppression evidence in the primary auditory cortex during movement (Schneider *et al*., 2014), we investigated whether similar suppressive effects can be seen at the level of the auditory brainstem and whether there are interactions with attention. The Frequency Following Response (FFR) to a 440Hz sine tone was measured while participants engaged in two finger tapping tasks equated in motor behavior but varying in attentional demand. Spectral peak analysis of the FFRs revealed decreased amplitude at 440 Hz for the tapping task that required more attention, suggesting that rather than a general suppression of the peripheral auditory pathway during motor behavior, the interaction between the motor system and the auditory brainstem is mediated by attentional networks that are involved in allocating resources to various sensory networks. [This work was supported in part by ONR grant DoD/ONR N00014-12-1-0850.]

3aPP13. Cortical dynamics of spatial and non-spatial auditory selective attention. Yuqi Deng, Hannah Goldberg, Barbara Shinn-Cunningham, and Inyong Choi (Boston Univ., 677 Beacon St., CompNet, Boston, MA 02215, vydeng@bu.edu)

Auditory selective attention suppresses processing of task-irrelevant stimuli, and it is crucial for effective communication in social settings. Previous studies showed that space- and pitch-based auditory attention engages different neural networks. However, the cortical dynamics underlying spatial and non-spatial auditory attention are unclear. Since accumulating evidence suggests that selective suppression is related to alpha band oscillation (8–14 Hz), we examine the spatial and non-spatial attentional modulation of alpha oscillation power as well as event-related potential (ERP) and behavioral performance. Using Electroencephalography (EEG) in humans, we compare behavior and physiological measures during *focused* attention (where listeners maintain focus on one “target” stream) and *broad* attention (where listeners are prepared to switch attention to a “super-target” stream which may or may not appear after the “target” stream) in spatial and non-spatial settings. We find that spatial attention shows overall stronger alpha power and different distribution in the central-parietal cortex. We also find that the monitoring cost of *broad* attention is higher in spatial attention, which is supported by a stronger ERP modulation. Our findings help elucidate the cortical dynamics involved in spatial and non-spatial auditory attention.

3aPP14. Age-related differences in auditory cortical representations of spatial cues. Erol J. Ozmeral, David A. Eddins, and Ann C. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210, Tampa, FL 33612, eozmeral@usf.edu)

Converging evidence indicates that binaural processing declines with age and is impaired further by age-related hearing loss. Research in young, normal-hearing adults indicates that spatial coding is governed by an opponent-channel (OC) mechanism. Under the OC model, the sensitivity to changes or shifts in perceived lateralization can be predicted based on the direction and magnitude of the shift, where large, outward shifts are predicted to cause a greater neural response than small and/or inward shifts. It is possible that age-related changes in spatial coding reflect changes in the OC mechanism, perhaps due to reduced neural inhibition or a more general reduction in temporal precision. Cortical event-related responses elicited by both ITD and ILD changes were measured using a 64-channel montage during a passive, continuous change-detection paradigm. Source localized cortical activation patterns were analyzed with regard to degree of lateralized shift, direction of shift (toward or away from perceived midline), and listener group. Results are consistent with an OC mechanism of spatial coding for both younger and older adults. Behavioral data from the same subjects are reported for correlation analyses with neural data. The present study provides a foundation for understanding the cortical dynamics of age-related changes in spatial tuning.

3aPP15. Cortical dynamics during spectrotemporal processing as indexed by directed functional connectivity. Ann C. Eddins and David A. Eddins (Commun. Disord. & Sci., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, aeddins@usf.edu)

Neural processing in the cortex plays a key role in our ability to discriminate changes in acoustic features important for understanding speech. Converging evidence demonstrates functional asymmetries between hemispheres such that the left hemisphere (LH) typically shows greater sensitivity to temporal changes while the right hemisphere (RH) shows greater sensitivity to spectral change. Most human studies to date have evaluated static differences in hemispheric activation with little attention given to potential dynamic changes within and across hemispheres during stimulus processing. The present study quantifies changes in the cortical representation of spectral (0.5 and 2.0 cycles/octave), temporal (2, 8, 32, and 64 Hz), and spectro-temporal (combination of both) modulation frequency. Cortical event-related responses were measured using a 64-channel montage during a passive, continuous change-detection paradigm in young, normal-hearing listeners. Individual and group data were preprocessed and source-localized for all conditions, with regions of interest (ROI) defined on group data. To determine whether activity in one ROI dynamically influences another during stimulus processing, Granger causality modeling was used to estimate directed functional connectivity. Preliminary results show robust directional processing that may originate in one hemisphere (i.e., spectral processing in RH or temporal processing in LH) but dynamically engages cortical regions within and across hemispheres.

3aPP16. The initial head gesture and coordinate system in anthropometric parameters measurement for head-related transfer functions customization. Guangzheng Yu, Yingyang He, and Bosun Xie (South China Univ. of Technol., Wushan Rd. 381#, Tianhe District, Guangzhou, Guangdong 510640, China, scgzyu@scut.edu.cn)

Head related transfer functions (HRTFs) depend on the sound source position and individualized anthropometry of subject. Accordingly, HRTF measurement usually accompanies with head and pinna-related anthropometric measurement on corresponding subjects. Based on a baseline database of HRTFs and anthropometric parameters, individualized HRTFs of new subject can be customized from anthropometric measurement (Zotkin, *et al*., 2003). Obviously, the accuracy of anthropometric measurement influences the performance of customized HRTFs. However, the measurement of some pinna-related anthropometric parameters, such as the pinna rotation angel and pinna flare angle, is sensitive to the initial head gesture and coordinate system. In present work, the Frankfurt plane was adopted to initialize the head gesture under the same coordinate system so as to ensure the consistency of anthropometric and HRTF measurement. The suggested method is adopted to measure the head and pinna-related anthropometric parameters of 56 subjects, yielding reasonable and consistent results. [This work was supported by the National Natural Science Foundation of China, Grant No. 11104082.]

3aPP17. Front-and-Overhead Energy Ratio and immersive sound field rendering with height channels. Sungyoung Kim and Mark J. Indelicato (ECT Eng. Technol., Rochester Inst. of Technol., ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com)

In a discrete channel-based music reproduction, height channels are required to completely manipulate increased immersiveness and enhanced appropriateness for a realistic three-dimensional (3D) sound field. Previous subjective evaluation results showed that the configuration of four height loudspeakers significantly changed perceived immersiveness and appropriateness, and the listeners preferred height-loudspeaker configurations with being "frontal" and "full." Subsequent analysis failed to determine a prediction model that could account for the variation of the listeners' perceptual responses using conventional physical parameters including inter-aural difference and coherence. In this paper, the authors proposed a new metric-the ratio between sound energy from front and overhead directions-as a physical parameter for the prediction model. A coincident pair of bidirectional microphones (one facing front and the other facing overhead) measured acoustical impulse responses (IRs) at the listening position for eight configurations of four height channels. Subsequent analysis measured the Front-and-Overhead Energy Ratio (FOER) between two IRs from the eight configurations and found that the ratios were highly correlated with the listeners' subjective rank data on perceived immersiveness and appropriateness (r = -0.8037). The result implicated that the appropriated configuration of height channels is a critical factor to render a convincing and immersive 3D sound field due to its direct influence on the FOER.

3aPP18. Efficient algorithm and localization experiment on spherical microphone array recording and binaural rendering. Yu Liu, Bosun Xie, Haiming Mai, and Jiayan Chen (South China Univ. of Technol., Wushan Rd. 381., Tianhe District, Guangzhou 510640, China, phbsxie@scut.edu.cn)

Spherical microphone array recording and binaural rendering (SMABR) is a novel spatial sound technique, which records spatial information of sound field, transfers to dynamic binaural signals, and then renders via headphone. To improve the computational efficiency of SMABR, the present work proposes a PCA-based (principal component analysis) algorithm for dynamic binaural synthesis from the beamforming outputs of spherical microphone array, in which only 33 pairs of sharable filters are required. Incorporated the proposed algorithm, a PC-based SMABR system consisting of a 64-channel spherical microphone array, an electromagnetic head-tracker, and a headphone is established. A virtual source localization experiment is carried out to evaluate the system and algorithm. Results indicate that the system yields reasonable localization performance within a target virtual source region near the horizontal plane. However, larger localization error is also observed for high elevation above 30 degrees or low elevation below -30 degrees. Localization error at high or low elevation is caused by the spatial aliasing error of spherical microphone array at high frequency, which spoils the spectral cue for elevation localization. Therefore, further improvement on spherical microphone array recording is needed. [Supported by the National Natural Science Foundation of China, Grant No.11174087.]

3aPP19. Neural network based speech enhancement applied to cochlear implant coding strategies. Tobias Goehring (ISVR, Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, T.Goehring@soton.ac.uk), Federico Bolner (Cochlear Technol. Ctr. Belgium, Mechelen, Belgium), Jessica J. M. Monaghan (ISVR, Univ. of Southampton, Southampton, United Kingdom), Bas van Dijk (Cochlear Technol. Ctr. Belgium, Mechelen, Belgium), Jan Wouters (ExpORL, KU Leuven, Leuven, Belgium), Marc Moonen (ESAT, KU Leuven, Leuven, Belgium), and Stefan Bleeck (ISVR, Univ. of Southampton, Southampton, United Kingdom)

Traditionally, algorithms that attempt to significantly improve speech intelligibility in noise for cochlear implant (CI) users have met with limited success, especially in the presence of a fluctuating masker. Motivated by previous intelligibility studies of speech synthesized using the ideal binary mask, we propose a framework that integrates a multi-layer feed-forward artificial neural network (ANN) into CI coding strategies. The algorithm decomposes the noisy input signal into time-frequency units, extracts a set of auditory-inspired features and feeds them to the ANN to produce an estimation of which frequency channels contain more perceptually important information (higher signal-to-noise ratio, (SNR)). This estimate is then used accordingly to suppress the noise and retain the appropriate subset of channels for electrical stimulation, as in traditional N-of-M coding strategies. Speech corrupted by various noise types at different SNRs is processed by the algorithm and re-synthesized with a vocoder. Evaluation has been performed in comparison with the Advanced Combination Encoder (ACE) in terms of classification performance and objective intelligibility measures. Results indicated significant improvement in Hit—False Alarm rates and intelligibility prediction scores, especially in negative SNR conditions. Findings suggested that the use of ANNs could potentially improve speech intelligibility in noise for CI users and motivated subjective listening experiments that will be presented together with the objective results.

3aPP20. Effects of phase difference on the binaural perceiving intensity. Sanmun Kim and Young H. Kim (Appl. Acoust. Lab, Korea Sci. Acad. of KAIST, 105-47, Baegyanggwanmun-ro, Busanjin-gu, Busan 614-100, South Korea, physicool13@gmail.com)

It is well known fact that human localize the direction of sound source in transverse plane according to interaural time difference in low frequency and interaural level difference on high frequency. In this research, we tested possibility for interaural time difference to be interpreted as interaural level difference when it is perceived by brain. To test the hypothesis, we used sinusoidal wave and commercial music to study the effect of interaural time difference on the perception of loudness. Each sound sample was regulated to have interaural time difference and given to 9 subjects. Subjects were asked to answer which side was louder. For sinusoidal waves, more than 60% of subjects answered side with earlier sound arrival was louder. The effect of interaural time difference on interaural level difference was larger when it interaural time difference was given to lower frequencies. Music samples were regulated with 100 and 500 interaural time difference and 100% of subjects answered side with earlier sound arrival was louder. Results from the experiments imply the possibility for interaural time difference interpretation by interaural level difference on localization of sound sources.

3aPP21. Our musical brain: Uncovering the neurophysical activation mechanism behind human perception of acoustics. Garrett W. Arosemena Ott, Tyler Blazey, Anish Mitra, Abraham Z. Snyder, and Marcus E. Raichle (Washington Univ., 6515 Wydown Boulevard, St. Louis, MO 63105, g.ott@wustl.edu)

If, in fact, neuronal activity patterns in the brain are dominantly driven by external forces and environmental stimuli (Hasson *et al.* 2004), rather than individual variation, music, and certain musical features, should elicit directly correlated neuronal responses across specific structures of the brain—effects as strong as that examination and analysis of brain activity in one subject could accurately predict activations in another (brain). Importantly, locating areal responses and activity patterns to musical stimulation could uncover novel information as to brain network functions, as well as yield insight into the neural foundations of the creative mind. Subjects, comprised of highly trained (HT) classical musicians (St. Louis Symphony) undergo functional Magnetic Resonance Imaging (fMRI) blood-oxygen-ation-level-dependent (BOLD) scanning to map neuronal activity triggered by intervals of music (Haydn Symphony No. 3), silence, and noise. Observed activation included the auditory, posterior cingulate, and visual cortices. The posterior cingulate cortex, a central node in the brain's Default Mode Network, has been strongly implicated in associations with episodic memory retrieval, working memory performance, human awareness, and several intrinsic control networks. The visual cortex is responsible for processing visual information; such activation may affirm the neuro-physical manifestation of HT performers' staple sensualization of music—music experienced as color, flavor, feeling, mood, etc.

3aPP22. Robust analysis of sound field reproduction by ambisonics based on singular-value decomposition. Dan Rao and Bosun Xie (Acoust. Lab.,School of Phys.,South China Univ. of Technol.,Tianhe district, Guangzhou, Guangdong 510641, China, phdrao@scut.edu.cn)

Ambisonics is a series of spatial sound systems with flexible loudspeaker configuration, which aims at reconstructing physical sound field in local region. Robustness of Ambisonics reproduction, which reflects the sensitivity of reproduced sound field to small errors, such as loudspeaker gain mismatch, is an important performance of sound reproduction. Condition number is generally used as an index to evaluate the robustness on whole, but this is insufficient because the robustness of reproduced sound field may be related to the target reproduction direction. To gain insights into the detail feature of robustness in Ambisonics reproduction, a method based on singular-value decomposition (SVD) is proposed to analyze reproduced sound field. The results indicate that the small singular values of transfer matrix and the proportion of Ambisonics coded vector projecting to the corresponding singular vectors dominate the robustness of Ambisonics reproduction. The robustness becomes worse only when such proportion of Ambisonics coded vector projection have relatively large energy. The proposed method is validated by simulating the sound pressure errors due to random mismatch of loudspeaker signals gain in some horizontal and spatial loudspeaker configurations. [Work supported by the National Natural Science Foundation of China, Grant No.11174087, and the State Key Lab of Subtropical Building Science, South China University of Technology, Grant No. 2014KB23.]

3aPP23. Interaural time and level interaction under free-field conditions. Brad Rakerd, Eric J. Macaulay, and William M. Hartmann (Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, rakerd@msu.edu)

Listeners judged the azimuthal location of sine tones presented in free field from an array of 25 small loudspeakers uniformly spaced over 180 degrees in the forward half of the horizontal plane. Probe microphones in the listener's ear canals recorded the tones, and transaural synthesis was used to reverse the interaural time difference (ITD) or the interaural level difference (ILD) across the mid-sagittal plane. Alternatively, the synthesis maintained the natural, consistent ITD and ILD. In the frequency range of 1000 Hz and below, localization judgments reflected the ITD, even as the interaural phase difference (IPD) surpassed 180 degrees with increasing azimuth or frequency, as long as the ILD and ITD had the same sign. In the reversed condition, and with small IPD, judgments reflected a compromise between ITD and ILD, with the ILD weight increasing with increasing frequency. But when the IPD exceeded a critical angle (about 120 degrees of phase) the influence of the ITD dramatically changed: Above 500 Hz, the localization judgments were consistent with ITD from a slipped cycle. At 500 Hz, judgments became chaotic. No critical angle occurred at 250 Hz. These conclusions are relevant in assessing the roles of interaural differences in sound localization. [Work supported by the AFOSR.]

3aPP24. The test-retest reliability of measurement procedures on most comfortable loudness levels for pure-tones. Cheng-Yu Ho (Holistic Education Ctr., Mackay Medical College, No. 46, Sec. 3, Zhongzheng Rd., Sanzhi Dist., New Taipei City 25245, Taiwan, swellfishyu@gmail.com), Pei-Chun Li (Dept. of Audiol. and Speech-Lang. Pathol., Mackay Medical College, New Taipei City, Taiwan), and Shuenn-Tsong Young (Holistic Education Ctr., Mackay Medical College, New Taipei City, Taiwan)

This study aimed to evaluate test-retest reliability of the forced-choice paired-comparison measurement procedures on most comfortable loudness levels (MCL) for pure-tones. Previous studies indicated that the MCL for pure-tones may be a range of level instead a fixed level, since the fixed-level MCL tends to be more inconsistent than a range of MCL. The ascending and descending measurement procedures were mostly used in the MCL for pure-tones, however the lower MCL obtained from ascending procedure, and higher MCL got from descending procedure, and since the low test-retest reliability of ascending and descending procedures, there are no well-established measurement procedures of MCL for pure-tones. The ascending and descending forced-choice procedure of MCL for speech might provide respectively 83% and 84% test-retest reliability, but the test-retest reliability measurement procedures of MCL for pure-tones have not been known. Therefore, this study proposed a new forced-choice paired-comparison measurement procedure on MCL for pure-tones, and evaluated the test-retest reliability of this proposed procedure. Normal hearing subjects are recruited to conduct this experiment, and evaluated by Pearson correlation. This experiment is ongoing, and the data collection would be finished before the presentation on the 170th Meeting of the Acoustical Society of America.

Session 3aSA

Structural Acoustics and Vibration, Signal Processing in Acoustics, and Physical Acoustics: Nonlinear Techniques for Nondestructive Evaluation

Brian E. Anderson, Chair
Dept. of Physics and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602

Invited Papers

8:00

3aSA1. Stress and strain monitoring in metals and alloys by guided acoustic waves with the aid of anharmonic effects and stress stiffening. Julian Grill and Wolfgang Grill (ASI Analog Speed Instruments GmbH, Burgweg 8, Koenigstein im Taunus, Hessen 61462, Germany, jg@analogspeed.de)

The basic theoretical features including anharmonic effects allowing structural health monitoring and the detection of stress and strain by guided acoustic waves are exemplified. Experimental results for stress and strain monitoring covering the linear elastic regime and the regime of plastic deformation are demonstrated for brass, copper and a standard aluminum alloy. Practically oriented applications, including also the monitoring of structural defects, load, weakening of structural components by overload and vibrations, and the detection of temperature to compensate for temperature dependent effects influencing structural health monitoring by acoustic waves, are presented for applications involving sections and components of aircrafts.

8:20

3aSA2. Nonlinear ultrasonic waves for structural monitoring: Thermal stress measurement and guided wave management. Francesco Lanza di Scalea (Structural Eng., Univ. of California San Diego, 9500 Gilman Dr., MC 0085, La Jolla, CA 92093, flanzadi@ucsd.edu), Claudio Nucera (Deutsche Bank, La Jolla, CA), and Simone Sternini (Structural Eng., Univ. of California San Diego, La Jolla, CA)

This presentation will cover two aspects of ultrasonic nonlinear wave propagation in solids. First, a new model is proposed to justify the existence of wave nonlinearities in constrained solids subjected to thermal excursions. This problem is solved on the basis of the interatomic potential of the solid that indicates a “residual” strain energy, due to the prevented thermal expansion, that is at least cubic as a function of strain. This study finds applications in the monitoring of thermal stresses in buckling-prone structures, such as continuously welded railroad tracks and pipelines. Experimental tests conducted on railroad tracks with realistic support will be also presented. Second, in the case of waveguides the efficiency of nonlinear ultrasonic testing based on higher-harmonic generation strongly relies on the correct identification of favorable combinations of primary and resonant double-harmonic nonlinear wave modes. This presentation will identify these combinations of wave modes in complex waveguides by extending the classical Semi-Analytical Finite Element formulation to the nonlinear regime, and implementing it into a highly flexible commercial Finite Element code. The proposed algorithm is benchmarked for four case-studies, including a railroad track, a viscoelastic plate, a composite quasi-isotropic laminate, and a reinforced concrete slab.

8:40

3aSA3. Nonlinear time reversal signal processing techniques applied to acousto-mechanical imaging of complex materials. Serge Dos Santos (Inserm U930 “Imaging and Brain, INSA Ctr. Val de Loire, 3, Rue de la Chocolaterie, Blois, Centre-Val de Loire F-41034, France, serge.dossantos@insa-cvl.fr), Zuzana Dvorakova (Inst. of ThermoMech. AS CR, Prague, Czech Republic), Michael Caliez (LMR, INSA Ctr. Val de Loire, Blois, France), and Zdenek Prevorovsky (Inst. of ThermoMech. AS CR, Prague, Czech Republic)

Recent ten years have seen considerable development of experimental techniques for improving nonlinear NDT methods and harmonic imaging derived from Nonlinear Elastic Wave Spectroscopy (NEWS). Diagnostic ultrasonic imaging based on higher harmonics yields, among others, a better resolution in view of the decreased wavelength in comparison with the fundamental. Furthermore, using symmetry invariance, nonlinear Time Reversal (TR) and reciprocity properties, the classical NEWS methods are supplemented and improved by new excitations having the intrinsic property of enlarging frequency analysis bandwidth and time domain scales. The purpose of this paper is to present the extension of TR-NEWS for skin aging characterization using the Nonlinear Time Reversal signal processing tool known to localize, in a complex medium, sources of nonlinearity potentially responsible of complex material aging. Linear and nonlinear behavior of skin elasticity is measured locally thanks to an acousto-mechanical loading of the skin conducted with INSTRON loading machines specifically optimized for biomaterials. Hysteresis behavior coming from the complex loading of the skin has been identified with PM-space statistical approach, usually associated to aging process in NDT. Phenomenological hysteretic parameters extracted will be presented and associated to standard parameters used for skin characterization.

9:00

3aSA4. Detection and localization of microcracking in polymer concrete using coda wave interferometry (CWI) at resonance.
Charfeddine Mechri, Mourad Bentahar (Acoustics, LAUM, Laboratoire d'acoustique de l'université du Maine, Ave. Olivier Messiaen, Le Mans 72000, France, charfeddine.mechri@univ-lemans.fr), Souad Toumi (Wave & Acoust., Haouari Boumdiene Univ., Laboratoire de Physique des Matériaux, Algers, France), Fouad Boubneider (Wave & Acoust., Haouari Boumdiene Univ., Laboratoire de Physique des Matériaux, Algers, Algeria), and Rachid El Guerjouma (Acoutics, LAUM, Laboratoire d'acoustique de l'université du Maine, Le Mans, France)

Nonlinear resonance and Coda Wave Interferometry (CWI) proved their potential to detect the evolution of a structure, namely, in the case of damage. Nevertheless, these techniques do not allow localizing defects in a structure. CWI may allow this localization through heavy statistical algorithms. In this case, the main problem remains the real time monitoring on one hand. On the other hand, in the case of highly scattering media, the amplitude of the acoustic pulse may cause the Coda signal to be affected by conditioning/relaxation effects. Moreover, Coda signal is very sensitive to experimental conditions, namely, to temperature. In order to overcome these inconveniences, we propose a method based on a comparative study of Coda signal contents in rest state and under a weak (linear) vibration. In this communication, we report the results obtained on a polymer concrete specimen excited under different resonance modes and for different dispositions of the sample which defect is not isotropic on which we apply simultaneously through transmission experiments in order to monitor changes in the coda signal.

9:20

3aSA5. Nonlinear ultrasonic technique for closed crack detection.
Kyung-Young Jhang (School of Mech. Eng., Hanyang Univ., 204 Eng. Ctr. Annex, 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, kyjhang@hanyang.ac.kr) and Hogeon Seo (Dept. of Mech. Convergence Eng., Hanyang Univ., Seoul, South Korea)

The detection of cracks at the early stage of fracture is important in industrial structures in order to guarantee their structural safety. Ultrasound has been widely utilized in the field of nondestructive testing of materials. However, most of these conventional methods using ultrasonic characteristics in the linear elastic region are mostly sensitive to opened cracks but much less sensitive to such closed cracks. The nonlinear ultrasonic technique (NUT) based on the contact acoustic nonlinearity (CAN) has been considered as a promising method for the closed crack detection. However, most of the previous studies were limited to the modeling of the second-order harmonic wave generation at contacted interfaces and its verification by testing artificially contacted interfaces in the through-transmission method. In this study, we investigated experimentally the contact acoustic nonlinearity at a real crack by using the measurement system constructed in the reflection mode that permits the transducers to access the only single side of a test structure. Results showed that the magnitude of the second-order harmonic wave represented the existence of the closed area clearly and that the crack sizing performance was greatly improved by the combination of the linear and nonlinear ultrasonic techniques.

9:40–10:00 Break

10:00

3aSA6. Pulse inversion and scaling subtraction signal processing for nonlinearity based defect detection.
Koen Van Den Abeele, Jan Hettler, Morteza Tabatabaeipour, and Steven Delrue (Dept. of Phys., KU Leuven - Kulak, E. Sabbelaan 53, Kortrijk 8500, Belgium, koen.vandenabeele@kulak.be)

When seeking out evidence for nonlinear behavior, various signal processing techniques can be applied for the comparison of two signals, one being a slight distortion of the other. For instance, the pulse inversion technique compares the responses to two out-of-phase excitation signals. Alternatively, one can compare the response at a finite (nonlinear) excitation amplitude to a scaled response at a very low (linear) excitation, as performed in the scaling subtraction technique. In this report, several examples are given in which these nonlinearity based signal processing techniques are used in practice to visualize damage features in solids. In view of kissing bond defect detection in friction stir welds, the pulse-inversion method was employed in a contact pitch-catch mode using a chirp signal. B-scan spectral heat maps obtained after pulse inversion allow to easily identify and size damage zones along the weld path. Second, the scale subtraction technique will be illustrated in combination with an ultrasonic sparse array SHM system to detect damage locations (impacts and delaminations in CFRP plates) without the knowledge of baseline signals taken on an intact specimen. Finally, we show that the phenomenon of Local Defect Resonance (LDR) can be facilitated and validated using the scaling subtraction technique.

10:20

3aSA7. Elasticity Nonlinear Diagnostic method for crack detection and depth estimation.
Pierre-yves Le Bas, Brian E. Anderson, Marcel Remillieux (Geophys. group, EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, pylb@lanl.gov), Lukasz Pieczonka (2. AGH Univ. of Sci. and Technol., Krakow, Poland), and Timothy J. Ulrich (Geophys. group, EES-17, Los Alamos National Lab., Los Alamos, NM)

In some Non-Destructive Evaluation techniques, detecting a crack is only the first step. Some applications like monitoring of used nuclear fuel canister also requires the characterization of the orientation of the cracks if any exist. Here, we show experimental results that expand the Time Reversal Elasticity Nonlinear Diagnostic method (TREND) to obtain the crack depth and orientation information. TREND is based on using time reversal to focus energy at a point of interest and then quantifying the nonlinearity at this point. By varying the frequency of the focal signal, one can interrogate different depth of the material, therefore getting access to information about the penetration depth of a crack. We will also show how Time Reversal can be used to select the orientation of the focus wave to any desired direction and how this, in turn, allows measuring different nonlinear responses according to this orientation and how this could be used to determine the orientation of cracks. [This work was supported by the US Department of Energy via the used fuel disposition campaign of the nuclear energy program.]

10:40

3aSA8. Model-based nonlinear guided wave approach for nondestructive evaluation in solid media. Younho Cho (School of Mech. Eng., Pusan National Univ., 10511, San 30, Jangjeon-dong, Geumjeong-gu, Busan 609-735, South Korea, mechcyh@pusan.ac.kr) and Weibin Li (Dept. of Aeronautics, Xiamen Univ., Xiamen, China)

The measurement of acoustic nonlinear response of guided wave propagation has been explored as a promising tool for early detection of micro-damages. Considering the high sensitivity of the nonlinear ultrasonic approach and the great advantages of the guided wave techniques, the nonlinear guided wave techniques have drawn significant attention. Frequency tuning and mode selection play a critical role for practical nonlinear guided wave testing. Among all the guided wave modes, it is of particular interest to find the more suitable modes for the improvement of efficiency. Analytical expressions and experimental observations of second harmonic generation of guided waves in an isotropic plate, are presented in this talk. Nonlinear parameters of guided waves are newly derived as a function of wave mode, geometric information of waveguides and frequency. Nonlinear features of various phase matched modes are discussed for the comparison of efficiency of second harmonic generation. The experimental results are consistent with the theoretical predictions. This study shows that physically based feature selection is essential for efficient generation of second harmonic guided waves. The feasibility of Lamb waves mixing for nonlinear ultrasonic test is also discussed in this work.

11:00

3aSA9. Soil-plate oscillator: A model for nonlinear acoustic landmine detection. Murray S. Korman (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu) and James M. Sabatier (National Ctr. for Physical Acoust., Oxford, MS)

A soil-plate oscillator (SPO) apparatus involves a cylindrical column of granular medium (sand, soil, and pebbles) supported by a thin circular elastic plate (acrylic) that is rigidly clamped to the bottom of a thick walled aluminum tube. The plate is air-backed. The soil column is driven from above by two subwoofers electrically connected to an amplified swept sinusoidal slowly varying chirp. In nonlinear tuning curve experiments, the resonant frequency decreases with increased amplitude—representing a softening in the nonlinear system. Hysteresis effects can be observed along with slow dynamic behavior (associated with mesoscopic nano-scale nonlinear elasticity). In two tone tests, numerous combination frequencies are observed in the surface vibration of the soil along with the sum frequency (typically 200 Hz) and second and higher harmonics. Soil surface vibrations and plate vibrations show similar effects when the soil layer is even 30 times the plate thickness. SPO results compare well when an inert VS 1.6 anti-tank mine is buried in a large concrete soil box. Nonlinear hysteresis models of the soil alone and for the soil-plate interaction are useful in understanding these results, since off-the mine vs. on-the mine tuning curves shapes behave significantly different, indicating some false alarms can be eliminated.

11:20

3aSA10. Characterization of air-coupled ultrasonic receivers for nonlinear Rayleigh wave nondestructive evaluation. David E. Torello (Mech. Eng., Georgia Inst. of Technol., 790 Atlantic Dr., Atlanta, GA 30332, david.torello@gmail.com), Jin-Yeon Kim (Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA), Jianmin Qu (Civil and Environ. Eng., Northwestern Univ., Evanston, IL), and Laurence J. Jacobs (Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA)

Nonlinear Rayleigh wave measurements can be used to determine damage precursors and measure microstructural material changes while only requiring one-sided access to a specimen. These measurements are extremely sensitive to coupling conditions between the specimen surface and the generating and receiving transducers used in propagation and measurement of the Rayleigh waves. Air-coupled detection offers many advantages to traditional contact techniques because it mitigates these experimental coupling concerns with a reasonable cost as contrasted with other non-contact methods. However, to use these devices to measure absolute nonlinearity in a specimen requires a detailed understanding of the transducers and the experimental setup. This work provides a combined numerical and analytical approach to understanding the received signal and uses this information to determine a framework for calculation of absolute measures of material nonlinearity.

Session 3aSC

Speech Communication: Various Topics in Speech Communication

Melissa M. Baese-Berk, Chair
Michigan State University, Oyer Center B-7, East Lansing, MI 48824

Contributed Papers

9:00

3aSC1. Human spoken language diversity and the acoustic adaptation hypothesis. Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu) and Christophe Coupé (Laboratoire Dynamique de Langage-CNRS, Lyon, France)

Bioacousticians have argued that ecological feedback mechanisms contribute to shaping the acoustic signals of a variety of species and anthropogenic changes in soundscapes have been shown to generate modifications to the spectral envelope of bird songs. Several studies posit that part of the variation in sound structure across spoken human languages could likewise reflect adaptation to the local ecological conditions of their use. Specifically, environments in which higher frequencies are less faithfully transmitted (such as denser vegetation or higher ambient temperatures) may favor greater use of sounds characterized by lower frequencies. Such languages are viewed as "more sonorous." This paper presents a variety of tests of this hypothesis. Data on segment inventories and syllable structure is taken from LAPSyD, a database on phonological patterns of a large worldwide sample of languages. Correlations are examined with measures of temperature, precipitation, vegetation, and geomorphology reflecting the mean values for the area in which each language is traditionally spoken. Major world languages, typically spoken across a range of environments, are excluded. Several comparisons show a correlation between ecological factors and the ratio of sonorant to obstruent segments in the languages examined offering support for the idea that acoustic adaptation applies to human languages.

9:15

3aSC2. Prosodic features of stance acts. Valerie Freeman (Univ. of Washington, Box 352425, Seattle, WA 98195, valerief@uw.edu)

While textual aspects of stance (attitudes/opinions) have been well studied in conversation analysis and computational models, acoustic-phonetic properties have received less attention. Recent work (2014 and 2015) has found that variations in prosodic measures (speech rate, vowel duration, pitch, and intensity) are correlated with stance presence and strength in unscripted speech, and stances with different discourse functions may be distinguishable by the shapes of their pitch and intensity contours. Building on these early findings, this presentation investigates prosodic properties of various stance-act types in spontaneous conversation (e.g., opinion-offering and soliciting, (dis)agreement, persuasion, rapport-building). The dataset contains over 32,000 stressed vowels from content words spoken by 40 speakers drawn from an audio corpus of dyads engaged in collaborative tasks. Speaker-normalized vowel duration, pitch, and intensity are automatically extracted from time-aligned transcriptions that have been hand-annotated for stance strength, polarity, and act type. Results show that changes in the prosodic measures combine to distinguish several notable stance-act types, including: weak-positive agreement, rapport-building agreement, reluctance to accept a stance, stance-softening, and backchanneling. Pitch and intensity contours over vowel duration are particularly illustrative, suggesting a future avenue in examining contours over whole stance acts.

9:30

3aSC3. Acoustic cues to the [j]-[i] distinction in American English. Zachary Jaggers (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, zackjaggers@nyu.edu)

Existing lexical items suggest that American English exhibits a [j]-[i] distinction (e.g., *pneumonia* [numonjə], *Estonia* [ɛstoniə]). This study tests if such a distinction can be experimentally elicited in both existing and new items and what acoustic cues most consistently convey it. A sentence reading task elicits the distinction by native speakers of American English using orthographically paired nonce names: 'y' stimuli (e.g., *Chobya*) expecting [jV] productions, 'i' stimuli (e.g., *Shabia*) expecting [iV] productions. Stimuli are controlled and diversified along the factors of place and manner of the preceding consonant and word position (initial vs. medial). Multiple acoustic factors of $[_{V]}$ sequences are measured and tested against each other as predictors of stimulus orthography, thus as cues to any elicited distinction, in a generalized linear mixed-effects model. Productions of 'y' stimuli are predicted by significantly earlier transition to the following vowel (represented by timepoint of the F2 maximum), lower F1, and lower intensity. This confirms the presence of the distinction and supports a constriction/height-based classification (Padgett 2008). A significant difference in F2 is not observed; these results are therefore not consistent with a classification of [j] as a coronal sound and [i] as dorsal (Levi 2008).

9:45

3aSC4. Does knowledge of probabilistic pronunciation patterns aid perception? Shinae Kang, Clara Cohen (Linguist, Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, cpccohen@berkeley.edu), and Rozina Fonyo (San Jose State Univ., Berkeley, CA)

Words which are probable in their morphological paradigms tend to have lengthened affixes (Cohen, 2014; Kuperman *et al.*, 2007). Here, we ask whether listeners use the pattern to aid perception. If so, paradigmatically probable words with lengthened affixes should be perceived more quickly than similarly lengthened improbable words. In two experiments (Experiment 1: phoneme monitoring; Experiment 2: lexical decision), we measured listeners' reaction time (RT) to 50 English verbs with [-s] suffixes (e.g., *looks*, *breaks*). Each suffix was adjusted in duration to either a normalized proportion of stem duration, shortened by 25% of the normalized duration, or lengthened by 25%. In Experiment 2, paradigmatic probability did not affect RT, but short words had slower RTs than normalized or lengthened words, suggesting that generally reduced suffix duration impedes perception. In Experiment 1, RT decreased with increased probability for the long condition, but not for the normalized condition. This suggests that a match between suffix length and paradigmatic probability facilitates perception. However, RT also decreased in the short condition, suggesting that when the stimulus was most difficult to perceive, listeners drew on general probabilistic information to aid perception. Our results further indicate that the effect of paradigmatic probability on perception is task-dependent.

10:00–10:15 Break

10:15

3aSC5. Extracting accent information in noise-vocoded speech in Japanese. Yukiko Sugiyama (Keio Univ., Hiyoshi 4-1-1, Kohoku-ku, Yokohama 223-8521, Japan, yukiko_sugiyama@mac.com)

This paper gives a preliminary report of a study that examined the perception of Japanese accent in noise-vocoded speech, where spectral information is substantially compromised while keeping the amplitude envelope intact. While the F0 is known to be the primary cue for accent in Japanese, it is not certain if secondary cues exist. Acoustic analyses conducted in previous studies show mixed results in this regard. The present study attempts to find non-F0 correlates of Japanese accent by conducting a perception study that used noise-vocoded speech. Twelve native speakers of Tokyo Japanese heard ten minimal pairs of final-accented and unaccented words (e.g., /hana*/ "flower" when accented vs. /hana/ "nose" when unaccented) embedded in a carrier sentence. The results obtained so far show that the listeners' accuracy exceeded chance level, suggesting that some acoustic information in the stimuli was present for the listeners to identify words. At the same time, relatively large individual differences were observed in their performance, indicating that some listeners were better at eliciting accent information than others. Implications of the results for the nature of Japanese accent and the perception of accent will be discussed.

10:30

3aSC6. Acoustic and perceptual correlates of subjectively rated sentence clarity in clear and conversational speech. Sarah H. Ferguson and Shae D. Morgan (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu)

Young adults with normal hearing and older adults with hearing loss performed subjective ratings of speech clarity on sentences spoken by all 41 talkers of the Ferguson Clear Speech Database. The sentences were selected from the CID Everyday Sentence lists and were produced under instructions to speak in a conversational manner and in a clear speaking style. A different set of 14 sentences was recorded in each style. Rated clarity will be compared between the two listener groups as well as among subgroups of talkers who differ in demographic and other characteristics. Clarity data will also be analyzed in conjunction with perceptual and acoustic data obtained in other investigations to reveal the relationship between vowel intelligibility and sentence clarity as well as the acoustic features that underlie perceived sentence clarity for different listener groups.

10:45

3aSC7. Developmental trajectory for perception of nonnative-accented sentences. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

The ability to recognize words under adverse listening conditions slowly develops throughout childhood. For example, children's speech perception in noise or reverberation does not reach maturity until adolescence. Much less is known about the developmental trajectory for children's word recognition under adverse listening conditions stemming from the talker, such as in cases of unfamiliar dialects or accents. To investigate development of word recognition with an unfamiliar accent, 5- to 15-year olds and young adults were presented with native- and Japanese-accented sentences in quiet and noise. Results showed that although 11- to 12-year olds' word recognition for the native in noise condition was similar to adults, the oldest children in the study (i.e., 14- to 15-year olds) did not demonstrate adult-like word recognition for the nonnative talker, with a large performance gap for the noise-added condition. Therefore, the developmental trajectory for word recognition with unfamiliar accents is quite protracted, similar to word recognition in noisy or reverberant environments. Because children's performance for the nonnative talker was depressed relative to adults even in quiet, children's difficulty perceiving unfamiliar accents is likely a consequence of cognitive-linguistic developmental factors or insufficient linguistic experience rather than sensory factors.

11:00

3aSC8. A comparison of speech enhancement methods to extract Lombard speech in an external noise field. Ghazaleh Vaziri, Christian Giguère (Rehabilitation Sci., Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, gvazi024@uottawa.ca), Hilmi Dajani (Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, ON, Canada), and Nicolas Ellaham (Rehabilitation Sci., Univ. of Ottawa, Ottawa, ON, Canada)

The tendency of talkers to increase their vocal effort in noise, known as the Lombard effect, depends on many factors including the type and level of noise. To investigate the influence of these different factors while wearing hearing protectors, the Lombard effect needs to be elicited by an external noise field (e.g., loudspeakers) rather than the traditional method (e.g., headphones). While the Lombard speech produced may be contaminated by the eliciting noise, the alterations in the talker's voice are more realistically accounted for by such methodology. The problem of recovering sound-field elicited Lombard speech has not been studied extensively. In this study, two noise suppression techniques, direct waveform subtraction and adaptive filtering, are used to recover the Lombard speech produced in simulated conditions using a manikin. To assess the performance of the two methods, the size of noise reduction is compared and basic speech characteristics (e.g., pitch and energy) and objective quality and intelligibility measures (e.g., SII and STI) are extracted from the clean and enhanced Lombard speech. Preliminary results show that the simulated Lombard speech could be accurately recovered, which is useful to extend to speech production in noise with real talkers.

Session 3aSP

Signal Processing in Acoustics: Random Matrix Theory in Acoustics and Signal Processing

James Preisig, Chair
JPAnalytics LLC, 638 Brick Kiln Road, Falmouth, MA 02540

Invited Papers

9:00

3aSP1. Application of random matrix theory to acoustic modeling and signal processing. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu)

Random matrix theory (RMT) characterizes the eigenvalues and eigenvectors of matrices composed of random entries [Bai/Silverstein, Springer, 2010]. Random matrices play an important role in a variety of acoustics and signal processing applications. For example, mode propagation through internal waves can modeled using random scattering matrices Hegewisch and Tomsovic showed that mode scattering can be analyzed using RMT [JASA, 2013]. In signal processing, the sample covariance matrix is an example of a random matrix that can be analyzed using RMT techniques. RMT differs from classical statistics because it derives results in the limit as both array size and number of measurements approach infinity. This type of asymptotic analysis is crucial because it facilitates the investigation of large dimensional matrices in scenarios with limited numbers of measurements. While results are derived in the infinite limit, many authors have shown that RMT provides useful predictions for finite size matrices derived from finite numbers of measurements. Research in RMT has grown substantially since the 1950s, and new theoretical results are rapidly emerging. This talk reviews several key results from the RMT literature and provides examples to illustrate the analysis of signal processing algorithms and acoustic propagation using these powerful methods.

9:30

3aSP2. Random matrix theory enabled performance analysis and algorithms for underwater signal processing. Raj R. Nadakuditi (Univ. of Michigan, 1012 Pontiac Trail, Unit 5, Ann Arbor, MI 48105, rajnrao@umich.edu)

Random matrices arise naturally in many undersea signal processing applications such as sonar and underwater acoustic communications. For example, the matrix formed by stacking a noisy time series of observations collected at a sensor array alongside each other is a random matrix. Random matrix theory provides a mathematical framework for reasoning about and understanding the structure in such noisy matrix-valued signals in an analogous manner to how Fourier analysis provides us a mathematical framework for reasoning about and understanding the structure in noisy vector valued signals. We highlight some recent breakthroughs in random matrix theory that have allowed us to predict the fundamental performance limits of weak signal detection, estimation and classification and discuss some recent successes where the theory has led to the development of powerful new algorithms for better estimating weaker signals than previously thought possible.

Contributed Papers

10:00

3aSP3. Probability distribution of multiple-target data snapshots applied to large-aperture array processing. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

We consider sonar array target detection and azimuth estimation in experimental scenarios consisting of large-aperture horizontal line arrays of hydrophones in the water column. In such scenarios, successful distinction of quiet targets from the background noise strongly depends on having sufficient independent data snapshots available for computation of adaptive beamforming weights. In this work, the discrete Fourier transform matrix is used for projecting the array data into a sparse domain, from which the theoretical probability density function of each (sparse-domain) datum can be obtained. It is shown that the sparse data exhibits a scaled chi-squared distributed magnitude, with a scale factor that depends on signal-to-background noise ratio and array size. Using this theoretical distribution and Monte Carlo simulations, minimum experimental requirements for target detection are quantified and highlight the tradeoff between data sample size, array aperture, and signal-to-noise ratio. In addition, a statistical test to reduce the number of false detections at the output of a beamformer is proposed and applied to simulated and experimental data from the Shallow Water Array Performance experiment.

10:15

3aSP4. The surprising sample covariance matrix: Unexpected characteristics and understanding them. Atulya Yellepeddi (Elec. Engineering/Appl. Ocean Phys. and Eng., MIT/WHOI, 77 Massachusetts Ave., Bldg. 36-683, Cambridge, MA 02139, atulya@mit.edu) and James Preisig (JPAnalytics, LLC, Falmouth, MA)

The Sample Covariance Matrix (SCM) is important to many problems in acoustic signal processing in both time-domain and frequency-domain processing. However, some careful analysis suggests that the SCM contains

a variety of surprises. For instance, not all elements of the SCM have the same error—the error of a particular element depends on the location of the element in the matrix and is different for SCMs of real-valued processes and complex-valued processes. Moreover, when the samples of the process used to compute the SCM are correlated, the sample covariance matrix behaves differently yet again. In particular, for frequency-domain sample covariance matrices, which are common in underwater acoustic communication and sonar signal processing, the SCM obtained using a tapped delay line is a better estimate of the covariance matrix than that obtained using independent samples—a most counterintuitive result. In this talk, we present a unified analysis technique using ergodic theory that predicts a variety of unexpected characteristics of the SCM, and explain the reasons behind some of them. Given the wide variety of algorithms that rely on the SCM, understanding this behavior is crucial to designing robust and accurate algorithms.

WEDNESDAY MORNING, 4 NOVEMBER 2015 RIVER TERRACE 2, 7:45 A.M. TO 10:40 A.M.

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Animal Bioacoustics: 50 Years of Underwater Acoustics under ASA

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943

Chair's Introduction—7:45

Invited Papers

7:50

3aUW1. Underwater Acoustics in 1950 and 1972. David L. Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu)

The National Academy (National Research Council) published "A Survey Report on Basic Problems of Underwater Acoustics Research" in 1950 and the Acoustical Society of America, in 1972, published a series of papers that were a 20-year review of various aspects of underwater acoustics. Between the two references, they provide a background of accomplishments and the perspective of scientists at that time, of the major issues in underwater acoustics. This paper is a summary of that information, and provides an "introduction" to the next 5 decades of research that is the focus of this special session. Though not the intent, a comparison of the research accomplished in later years with the proposed effort will provide a basis for further discussion.

8:10

3aUW2. Project MiMi—Miami-Michigan—The discovery of phase coherent acoustic transmission. Harry A. DeFerrari (Ocean Sci., Univ. of Miami, 1148 N E 89 St., Miami, FL 33138, hdeferrri@rsmas.miami.edu)

Project MIMI aimed to observe the temporal fluctuations of low frequency CW acoustics transmission. At the time, the conventional wisdom was that CW phase was randomized after transmission through the ocean. Experiments (transmission between ships) verified that finding and ray theoretical considerations confirmed that uncertainty in the depth of the upper turning point resulted in large variations in travel time—hence random phase. But, some classified work suggested that the ocean transmission was perhaps stable and coherent. John Stienberg experimental group at Miami installed a 400 Hz. bottom mounted directional source aimed at Bimini, Bahamas, but could barely detect the signal. Ted Birdsall provided signal processing gain (36 dB) with the Phase Coherent Demodulator (PCD) that lead to the discovery of phase coherence. The Miami—Michigan (MiMi) team resulted. John Stienberg found that phase variations had periodicities corresponding to surface wave, internal wave, tide and lunar periodicities. He concluded that by "Oceanographic Acoustics" one could make acoustic measurements of oceanographic variations. Birdsall introduces M-sequences ("Gain of CW with the resolution of a pulse") to observe pulse arrivals This approach served as a template for dozens of future fixed-system experiments by the basic research community up until the present time.

8:30

3aUW3. Sixty years studying wave propagation in random media at the Applied Physics Laboratory. Terry E. Ewart and Daniel Rouseff (Appl. Phys. Lab., Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, teewart@gmail.com)

Ocean acoustics has been a useful avenue for testing evolving theories for Wave Propagation in Random Media (WPRM). These theories generally assume that the index of refraction statistics are stable in space and time, an assumption proven reasonably true in the deep ocean for acoustic paths away from boundaries. In the present work, results from 60 years of theoretical and experimental WPRM research at the University of Washington's Applied Physics Laboratory (APL) are reviewed. The first experiment was performed in 1959 to test theories for amplitude fluctuations based on the Born approximation. The Rytov approximation (from Russian literature) for calculating the log-amplitude fluctuations was also evaluated. Conclusion: neither applied. Experiments in 1971 and 1977 measured acoustic fluctuation statistics for an 18 km acoustic path at sonar-relevant frequencies, 2–13 kHz. A 1985 experiment under Arctic ice used 2–16 kHz signals over a 6 km path. These experiments are discussed together with theoretical issues based on the Moment Equation method to provide one viewpoint on the history of ocean acoustic WPRM. The following translation of Voltaire is appropriate: "The ancients when reasoning about physics without the enlightenment of experiments are like blind men explaining the nature of colors to other blind men."

8:50

3aUW4. ONR arctic acoustics 1978—1998. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu) and Peter N. Mikhalevsky (Leidos Corp, Arlington, VA)

The Arctic Program Office of the Office of Naval Research ten Arctic field programs from 1978–1994 under the visionary leadership of program managers Dr. G. Leonard Johnson and Dr. Tom Curtin. During this period, over ten ice camps in both the western Arctic (Beaufort Sea) and the eastern Arctic (Nansen and Pole Abyssal Plains were manned and four ice breakers served as platforms in the marginal ice zone (Fram Straits). Since the cost of the support logistics for Arctic field programs is so very high, these experiments were multidisciplinary and almost all had an acoustic component. Some of the highlights were transoceanic reverberation, seismic reflection and refraction, random channels for time and Doppler spreading, target detection, matched field processing,ocean acoustic tomography, seismicity, and ambient noise were among the many topics examined. There were also robust efforts advancing data acquisition. Large, two dimensional horizontal arrays with both cabled and "WIFI" telemetry, large vertical arrays, precision sensor navigation, and sophisticated remote instrumentation buoys were deployed. With the end of the "Cold War" the last field program was in 1994 and the ONR Arctic program eventually was disestablished. Now, the retreat Arctic ice cover and Arctic Ocean warming has reinvigorated ONR's interest in the Arctic and after two decades ONR field programs are planned for the near future.

9:10–9:25 Break

9:25

3aUW5. Nonlinear acoustics and the Acoustical Society of America. Thomas G. Muir (The University of Texas at Austin, Appl. Res. Labs., P.O. Box 8029, Austin, TX 78713, tgmuir@earthlink.net), David G. Browning (None, Kingston, RI), and Kenneth G. Foote (Woods Hole Oceanographic Inst., Woods Hole, MA)

The modern era of nonlinear underwater acoustics had its roots in a remarkable session at the 57^{th} ASA meeting at Providence RI in June, 1960, chaired by Westervelt, of Brown University, who also presented his remarkable new theory of the parametric acoustic array. This conceptual "device" enables highly directive low frequency sound to be created from the nonlinear interaction of two high frequency radiations of high directivity, which was confirmed in a model- tank study paper in this session by Brown Univ. Professor Robert Beyer and student Bellin. JASA has been the major worldwide forum for publications in nonlinear underwater acoustics, ever since, with much of the U.S. work done at the Navy Underwater Sound Laboratory and its successor, the Naval Underwater Systems Center, as well as the Applied Research Laboratories of the University of Texas at Austin. Topics have included parametric arrays, the nonlinear generation of harmonic radiations and transients, acoustic saturation, nonlinear transmitting and receiving sonar and imaging, and many practical applications to underwater sound and acoustical oceanography. Current and/or developing applications include underwater communications, backscattering measurements, penetration of narrow beams into sediments, sub-bottom profiling, and fish school quantification and classification. These and other topics are reviewed and recent implementations are described. [Work supported by ARL:UT Austin.]

9:45

3aUW6. Deep-water ocean acoustic propagation: Observations. Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu)

There is a rich history of long-range, low-frequency, deep-water ocean acoustic propagation measurements extending back to the discovery of the deep sound channel by Ewing and Worzel in 1944. Experiments up to the 1970s focused on measuring parameters in the sonar equation, including transmission loss and ambient noise, motivated in part by the development of the U.S. Navy Sound Surveillance System (SOSUS). These measurements generally used wideband explosive sources or narrowband transducers. Beginning in the 1970s, low-frequency, broadband transducers and vertical receiving arrays that could operate autonomously and be moored for extended periods were developed in connection with the new field of ocean acoustic tomography. These developments allowed individual multipaths and modes to be resolved and long time series collected, so that the fluctuations in the received signals could be quantified. At about the same time, advances in characterizing ocean internal wave and mesoscale variability provided information on the causes of the acoustic fluctuations that could be used in theoretical calculations. The study of deep-water propagation and the development of ocean acoustic tomography have been intertwined ever since. The applications of deep-water propagation now extend beyond military uses to ocean acoustic remote sensing (active and passive), communication, and navigation.

10:05

3aUW7. Shallow water experiments and results. James Lynch, Timothy F. Duda, Arthur E. Newhall, and Ying T. Lin (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

Shallow water acoustics has flourished over the past fifty years, but most especially over the last 25 years. After World War II, the Cold War between the USSR and the West focused the emphasis in ocean acoustics on deep, "blue water." After the Cold War waned in 1990, the emphasis changed to "brown water" coastal acoustics studies. In this paper, we will look at the advances over the past half-century in shallow water acoustics, with emphasis on the experimental results from acoustics, oceanography, marine geology, and marine biology.

Contributed Paper

10:25

3aUW8. Bottlenose dolphins direct sonar clicks off-axis of targets to maximize Fisher Information about target bearing. Laura Kloepper (Dept. of Biology, Saint Mary's College, 185 Meeting St. Box GL-N, Brown Univ.Dept. of Neurosci, Providence, RI 02912, laurakloepper@gmail.com), Yang Liu, and John R. Buck (Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Aiming the sonar beam directly at a potential target maximizes the detection information available to an echolocating animal. In contrast, estimating the angular location of a target relies on changes in the received echo spectrum due to the frequency dependence of the transmit beam pattern. The Fisher Information quantifies the available information on angular location in terms of the sensitivity of the received echo spectrum as a function of bearing. We calculated the Fisher Information for a dolphin's echolocation signal and determined the maximum angular location information occurs when the sonar beam is such that the target falls slightly off-axis. To compare the predicted beam aim to the actual beam aim of an echolocating dolphin, we recorded the echolocation signals of a bottlenose dolphin with a 16-element hydrophone array while the animal performed a target detection task. The dolphin consistently pointed its sonar beam 7 degrees away from the target, which is the beam aim that maximizes the Fisher Information for the dolphin signal. [Work supported by NSF and ONR.]

WEDNESDAY MORNING, 4 NOVEMBER 2015 GRAND BALLROOM FOYER, 9:00 A.M. TO 12:00 NOON

Session

Exhibit and Exhibit Opening Reception

The instrument and equipment exhibit is located near the registration area in the Grand Ballroom Foyer.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Exhibit hours are Monday, 2 November, 5:30 p.m. to 7:00 p.m., Tuesday, 3 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 November, 9:00 a.m. to 12:00 noon.

Coffee breaks on Tuesday and Wednesday mornings (9:45 a.m. to 10:30 a.m.) will be held in the exhibit area as well as an afternoon break on Tuesday (2:45 p.m. to 3:30 p.m.).

The following companies have registered to participate in the exhibit at the time of this publication:

Brüel & Kjær Sound & Vibration Measurement—www.bksv.com

Freudenberg Performance Materials—www.Freudenberg-pm.com

G.R.A.S Sound & Vibration—www.gras.us

PCB Piezotronics—www.pcb.com/

Sensidyne—www.sensidyne.com

Springer—www.Springer.com

Teledyne Reson—www.teledyne-reson.com

Session 3pAA

Architectural Acoustics: AIA CEU Course Presenters Training Session

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

All are welcomed, but TCAA membership and sign-in/sign-out attendance for the entire two hours of this workshop are required to qualify as an authorized presenter for this AIA/CES short course.

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. TCAA short course presentation material. K. A. Hoover (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects, called "Architectural Acoustics." An architect can earn one continuing education unit (CEU) by attending this short course, if it is presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, and finish treatments. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this workshop and membership in TCAA are required.

2:05

3pAA2. AIA/CES Provider registration and reporting requirements. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects. The TCAA short course is called "Architectural Acoustics" and attendance at this one-hour long course can earn an architect one continuing education unit (CEU) with HSW Credit (Health Safety and Welfare). This paper will cover the administrative requirements of the AIA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork, so that AIA members may receive credit for the course. Also, the manner in which the course is given is dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course.

Session 3pBA

Biomedical Acoustics: Therapeutic Ultrasound, Microbubbles, and Bioeffects II

Wayne Kreider, Chair
CIMU, Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

1:00

3pBA1. *In vivo* biodistribution of fluorescently tagged magnetic microbubbles for cavitation enhancement with real time passive acoustic mapping. Calum Crake, Robert Carlisle, Joshua Owen, Sean Smart, Christian Coviello, Constantin Coussios, and Eleanor P. Stride (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Previous work has demonstrated the potential of magnetically functionalized microbubbles to localize and enhance cavitation activity under focused ultrasound exposure *in vitro*. The aim of this study was investigate magnetic targeting of microbubbles for promotion of cavitation *in vivo*. Fluorescently labeled magnetic microbubbles were intravenously injected into a murine xenograft model. Cavitation was induced using a 0.5 MHz single element focused transducer at peak negative focal pressures of 0.1–1.0 MPa and monitored in real-time using simultaneous B-mode imaging and passive acoustic mapping. Magnetic targeting was found to increase the amplitude of the cavitation signal as compared with untargeted bubbles and a commercial ultrasound contrast agent. Post exposure magnetic resonance imaging indicated a correlation between cavitation activity and deposition of magnetic nanoparticles in the tumor volume. Magnetic targeting was similarly associated with an increase in the fluorescence intensity measured from the tumors following the experiments; the highest levels corresponding to the maximum cavitation signals. These results indicate a strong correlation between cavitation activity and increased uptake of both molecular species and nanoparticles in tumors; and that the effect can be enhanced by magnetic targeting.

1:15

3pBA2. Acoustic methods for modulating the cavitation initiation pressure threshold. Hedieh A. Tamaddoni, Alexander P. Duryea, and Timothy L. Hall (Univ. of Michigan, Biomedical Eng., 2109 Gerstacker, Ann Arbor, MI 48109, alavi@umich.edu)

The objective of this study is to develop tissue protection methods by modulating bubble cloud initiation pressure thresholds. Specifically, we investigated pressure thresholds to initiate cavitation bubble clouds by the shock scattering mechanism. Cavitation initiation displays a stochastic nature affected by existing nuclei populations in the medium. We hypothesized that by applying proper low pressure pulse sequences before and/or during histotripsy therapy, initiation pressure threshold and growth of cavitation bubble cloud could be modified. We applied histotripsy and cavitation suppressing pulses in both water and agarose gel for pulse repetition rates of 1, 10, and 100 Hz. Acoustic backscatter signals and optical imaging were used to detect and monitor initiation, maintenance, and growth of resulting cavitation bubble cloud. Results demonstrated that the use of cavitation suppressing pulses can increase the cavitation threshold by 20% in the targeted space. Furthermore, we showed these acoustic sequences could modify the shape and density of the bubble cloud. By applying the cavitation suppressing pulses, we were able to generate a dense cavitation bubble cloud in the focus while decreasing scattered cavitation in the peripheral zone.

1:30

3pBA3. Experimental and numerical evaluation of the effect of stone size on fracture by burst wave lithotripsy. Madeline J. Hubbard, Barbrina Dunmire (Ctr. for Medical and Industrial Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Krieder, Michael R. Bailey (Ctr. for Medical and Industrial Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St, Seattle, WA 98105, amax38@u.washington.edu)

Burst wave lithotripsy (BWL) is an experimental treatment to noninvasively fragment kidney stones using bursts of focused ultrasound. Preliminary simulations with a linear elastic model showed that resonance creates concentrated stresses, which may help predict locations of fractures in the stones. In this study, we aimed to demonstrate this correlation by comparing simulations to experimental data. Cylindrical stones of variable size (4–14 mm diameter, 10 mm length) made from BegoStone plaster were treated in a water bath for 10 min using a 170 kHz focused transducer at a focal pressure of 6.5 MPa. Locations of first fractures in the stones correlated well with the location of peak stress predicted in the linear elastic model. Simulated peak surface stress in the stones decreased as stone diameter increased, with the exception of a spike near $d = 12$ mm, which matches the half longitudinal wavelength in the stone. Experimentally, a corresponding overall increase in time to first fracture was observed with diameter, except for a drop at $d = 12$ mm. The results are encouraging that the model may help direct further optimization of BWL. [Work supported by NIH through DK043881, DK104854, EB007643, and NSBRI through NASA NCC 9-58.]

1:45

3pBA4. Fast passive cavitation mapping with angular spectrum approach. Costas Arvanitis, Nathan McDannold (Radiology, Harvard Med. School, Brigham and Women's Hospital, 221 Longwood, Rm. 514A, Boston, MA 02115, arvanitis.costas@gmail.com), and Gregory Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH)

Several emerging focused ultrasound (FUS) therapies harness the effects produced by acoustic cavitation. Passive acoustic mapping (PAM) can be used to characterize and visualize the microbubble oscillations and guide these procedures. Here, we propose and demonstrate that angular spectrum approach (ASA) can be used to back-project the passively recorded, by an array of receivers, diverging pressure waves generated from oscillating microbubbles to perform rapid PAM reconstructions. In the present work, acoustic cavitation is studied transcranially in non-human primates using an integrated ultrasound and MRI-guided clinical FUS system. In addition, from CT datasets, we also extract the skull acoustic properties and use them as inputs to numerical simulations. Using both the simulated and experimental data we validate the use of ASA to perform PAM reconstructions and compare its performance to time domain PAM. The experimental data demonstrate that ASA can be used to reconstruct frequency-selective PAM. Numerical simulations suggest that ASA reconstructions have the same resolution with time domain PAM, while the reconstruction time was 72 times faster for the same image dimensions. These results suggest that ASA-based PAM reconstructions can provide real-time passive cavitation mapping and, by extension, control over FUS procedures.

2:00

3pBA5. Acoustic levitation device for probing biological cells with high-frequency ultrasound. Brian D. Patchett (Phys., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, brian.d.patchett@gmail.com), Natalie C. Sullivan (Chemistry, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency ultrasound has been shown to be sensitive to changes in the cell cytoskeletal makeup during a variety of biological processes. It can therefore be used to study cellular responses such as T-cell activation and cancer metastasis. To date, obtaining cytoskeletal properties with high-frequency ultrasound has required the use of monolayer cell cultures. Ultrasonic signals from these cultures can be problematic due to the interference of reflections from the culture-plate well. The cell structure is also often deformed from its native state due to adhesion to the culture plate. The objective of this study was to develop an acoustic levitation device that creates a monolayer of cells suspended in a fluid in order to simulate their environment *in vivo*. By using a 250 kHz transducer, standing waves were generated in a suspension of polyethylene microspheres (53–63μm diameter) in distilled H_2O. Microspheres formed layers at the standing-wave nodes. Varying the shape of the transducer voltage waveform had a significant effect on layer thickness, with a square waveform creating thinner, more distinct layers than a sine waveform due to steeper pressure gradients at the nodes. Future work will entail creating a monolayer suspension of biological cells that can be probed with high-frequency ultrasound.

2:15

3pBA6. *In vivo* cavitation thresholds and injury observations related to burst wave lithotripsy. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan W. Cunitz, Yak-Nam Wang (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Donghoon Lee (Dept. of Radiology, Univ. of Washington School of Medicine, Seattle, WA), Mathew D. Sorensen, Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Oleg A. Sapozhnikov, Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Michael R. Bailey (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a new non-invasive approach for treating kidney stones that uses short bursts of ultrasound at low duty factors. An important difference between BWL and SWL relates to the potential for generating cavitation in the kidney that could cause injury. To study such cavitation behavior, 17 pig kidneys were exposed *in vivo* to BWL treatments of different amplitudes at frequencies of 170 and 335 kHz. Treatments lasted at least 10 min and were monitored in real time using ultrasound imaging (including B-mode and Doppler modalities) to determine threshold pressures for the onset of cavitation. After treatment, eight of the kidneys were perfusion fixed and scanned using MR imaging sequences designed to detect hemorrhagic injury. Results suggest that pressure thresholds exist below which cavitation detected by ultrasound imaging is very unlikely to occur; moreover, MR images indicate no injury or only minimal injury when cavitation was not detected. Observations at 170 kHz show a repeatable threshold pressure near 5 MPa; data at 335 kHz were less consistent but suggest a higher threshold between 6 and 8 MPa. [Funding support by NIH NIDDK P01-DK043881, R01-DK092197, T32-DK007779, K01-DK104854, R01-EB007643, R01-CA188654, and NSBRI through NASA NCC 9-58.]

2:30

3pBA7. Focused ultrasound-enhanced intranasal delivery of brain-derived neurotrophic factor. Hong Chen, Georgiana Z. Yang, Hoheteberhan Getachew, Camilo Acosta, Carlos S. Sánchez (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., PS 19-418, New York, NY 10032, hc2666@columbia.edu), and Elisa E. Konofagou (Dept. of Biomedical Eng. and Dept. of Radiology, Columbia Univ., New York, NY)

The therapeutic use of neurotrophic factors in the treatment of central nervous system diseases has been restrained by their low blood-barrier permeability and rapid degradation in the blood. Intranasal (IN) administration is a promising approach for delivering neurotrophic factors directly to the brain, bypassing the BBB. However, IN delivery has low efficiency and does not offer localized delivery to specific brain sites. The objective of this study was to demonstrate the feasibility of FUS-enhanced IN delivery of brain derived neurotrophic factor (BDNF) at a targeted location. BDNF was administered through IN route in wild-type mice ($n = 7$) followed by FUS sonication at the left caudaputamen in the presence of systemically circulating microbubbles. The contralateral right caudaputamen was used as control for IN delivery only. Immunohistochemistry staining was used to assess the distribution of BDNF and the bioactivity of BDNF in activating the downstream signaling. It was found that FUS enhanced the delivery efficiency of IN administered BDNF at the targeted region, where BDNF penetrated deep into the brain tissue instead of confining within the perivascular spaces as in the contralateral control side. Furthermore, the delivered BDNF reached sufficient concentration to activate the downstream signaling pathway.

2:45

3pBA8. Ultrasound-mediated drug release from nanoscale liposomes using nanoscale cavitation nuclei. Susan Graham, James Kwan, Rachel Myers, Christian Coviello, Robert Carlisle, and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

We have previously presented a liposomal formulation of mean size 140 nm, manufactured using DSPE, cholesterol, DSPC and DSPE-PEG at ratios of 65:25:3:7, which exclusively releases encapsulated doxorubicin in the presence of inertial cavitation nucleated by microbubbles (SonoVue®, Bracco) at peak rarefactional pressures in excess of 1.2 MPa at 0.5 MHz (Graham *et al.*, J. Controlled Release, 2014). However, the benefits of cavitation-sensitive liposomes small enough to pass through the leaky tumor vasculature can only be fully realized if they can be triggered by cavitation nuclei which are also small enough to extravasate into the tumor mass. In the present work, we demonstrate that liposomal release comparable to that mediated by SonoVue® microbubbles can be achieved using gas-stabilizing polymeric nanocups of mean diameter 400 nm, at peak rarefactional pressure amplitudes in excess of 1.5 MPa at 0.5 MHz or 4 MPa at 1.6 MHz. Mechanistically, we hypothesize that release occurs once a threshold peak shear rate is exceeded in the fluid surrounding the collapsing microbubble, thus exceeding the critical shear stress on the liposomal surface. This is confirmed experimentally by demonstrating a correlation between release and the maximum power of broadband acoustic emissions received by a passive cavitation detector.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Ning Xiang, Chair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Chair's Introduction—2:10

Invited Paper

2:15

3pED1. Propagation of acoustic education in space and time. Yang-Hann Kim (KAIST, Dept. of M.E., Sci. Town, Daejon-shi 305-703, South Korea, yanghann@kaist.ac.kr)

Sound is every where and any time. Human is always surrounded by sound. However, understanding sound is not always available to everyone. It's concepts start to be introduced from elementary school to college, but implementing sound for his/her purpose is always problematic. One reason why sound is so difficult to understand is because it is not visible. There have bee numerous attempts to make sound visible: using water, many candle lights, string, small particles, and many microphones. Another hurdle that we have to go over to really enjoy and use sound is to have a convenient tool for manipulating sound. However making one's own speaker cannot be tackled by using a home took box. However, we are about to make this dream come true: many good speakers can be found in relatively cheap price and tangible manipulation tools begin to be available. This paper introduce how these tools have been developed and can be used for education. Another issue associated with learning sound is how to make acoustic education available to anyone and anytime: distance education, using Internet can be a solution. Past experience will be introduced: college lecture, short course for industry, Internet lecture, and finally MOOC lecture for acoustics are compared. What their characteristics are, and how to use them are to be discussed.

WEDNESDAY AFTERNOON, 4 NOVEMBER 2015 GRAND BALLROOM 2, 1:00 P.M. TO 2:05 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Tessa Bent, Chair
Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Hot topics in underwater acoustics: Recent advances in underwater unexploded ordnance classification. Aubrey L. Espana, Kevin L. Williams, and Steven G. Kargl (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu)

For decades, broadband sonar has been a key tool in the underwater detection and classification of man-made targets. The DoD's recent focus to clean up discarded WWII-era munitions has pushed a flurry of research and advancements in the areas of target scattering, environmental noise modeling, signal processing, and automatic target recognition (ATR). It is the interdisciplinary nature of this

problem that makes it incredibly challenging, namely, the fact that the ability to classify an object is heavily influenced by the operational choices (grazing angle and vehicle altitude), surrounding environmental factors (target type, burial state, sea state, and bottom topography), and the processing tools (data products, feature-set, and ATR training/testing). This talk will outline the key steps underlying the process of trying to detect and classify underwater unexploded ordnance (UXO). It begins by pushing raw time domain data, which can be generated either experimentally or via models, through a set of signal processing tools to create data products (acoustic color and SAS images). These data products are fed through feature extraction processes, followed by ATR algorithms, in order to arrive at the final classification of an object. Examples to be presented include objects of interest to the remediation of underwater UXO. [Research supported by SERDP and ONR.].

1:25

3pID2. Seismic surveys and marine wildlife: Impacts, the lack thereof, and thoughts on managing both. Douglas P. Nowacek (Nicholas School of the Environment and Pratt School of Eng., Duke Univ. Marine Lab., 135 Duke Marine Lab Rd., Beaufort, NC 28516, dpn3@duke.edu)

Sufficient scientific data exist to conclude that seismic airguns used in geophysical exploration have a low probability of direct harm to most marine life, except at close range where physical injury is a real danger. Further, airguns in some conditions do not appear to disturb animals; however, in other conditions, they result in moderate to extreme behavioral responses and/or acoustic masking over large areas. Additionally, recent studies have reported the presence of seismic survey sound energy over ranges of ~4000 km. While the potential for effects have not even been investigated at such ranges, the presence of the signals must be taken into account when evaluating overall potential for impacts. Mitigation measures have historically focused on reducing immediate harm, but systematically measuring and understanding the full potential for impacts is an important aspect of any responsible development program. The European Union has recognized ocean noise as a pollutant and as an indicator of environmental quality under its Marine Strategy Framework Directive. Given this and the international and transboundary nature of noise from marine seismic surveys, their ubiquity, the presence of numerous other sources of ocean noise, and that incorporating acoustic disturbance into an understanding of population level consequences is progressing, a responsible path forward should focus on the creation of legally binding international commitments and standards for the management and minimization of noise.

1:45

3pID3. Hot topics in "cold" acoustical oceanography. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Acoustics is a vital tool for probing the ocean interior, the seafloor, and the sea surface. This talk on Acoustical Oceanography will focus on hot topics in cold places, specifically the Arctic. The rapid loss of sea ice in the arctic circle is leading to an increase in human activities, with implications for oil and gas exploration, fisheries, shipping, and tourism, to name a few areas. Current active and exciting research areas in Acoustical Oceanography include: (1) multipurpose acoustic networks in the Arctic that support the passive monitoring of underwater sound, and remote sensing activities such as acoustic tomography and thermometry, (2) using acoustics to track marine mammals and evaluate their responses to industrial noise, (3) the use of high frequency acoustics to detect and quantify oil layers beneath sea ice, and (4) the use of underwater ambient noise to study glacier dynamics and ice melting in arctic, glacial bay.

Session 3pSC

Speech Communication: Speech Style and Sociophonetics (Poster Session)

Benjamin Munson, Chair
Speech-Language-Hearing Sciences, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive SE, Minneapolis, MN 55455

Authors should be at their posters from 1:45 p.m. to 3:15 p.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 1:00 p.m. to 3:15 p.m.

Contributed Papers

3pSC1. Schwa reduction and the realization of /p, t/ in casual American English. Ellen Aalders and Mirjam Ernestus (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, Erasmusplein 1, Nijmegen, Gelderland 6525 HT, Netherlands, e.aalders@let.ru.nl)

This study focused on the realization of /p, t/ after schwa (e.g., *support, certificate*) in casual American English. We analyzed the realization of 523 tokens in the Buckeye corpus. Pronunciation variation proved substantial. Some plosives were realized as fricatives or were completely absent; others varied in the duration and voicing of the closure, burst presence, and voice onset time (VOT). Tokens of *suppose*(*d*) showed markedly more reduction than other words: schwa was more often absent (80% versus 28%) and VOT was much shorter (mean 19 ms versus 59 ms). We propose that *suppose(d)* has an additional lexical entry without schwa: *spose(d)*. It is typically assumed that if the schwa is lost, a following voiceless plosive retains the long VOT that is typical in stressed syllable-initial position (e.g., *support* pronounced as [s'p^{h}ort]). We compared VOT in our schwa words, excluding *suppose*(*d*), with that of 405 plosives occurring after /s/ (e.g., *sport*, *still*). Regression analyses showed that VOT was longer in syllable-initial position than after /s/ in consonant clusters (mean 16 ms). Schwa absence did not affect VOT and neither did the duration of schwa when present. This suggests that the reduction processes affecting vowels need not change the realization of consonants.

3pSC2. Influence of clear speech on the word intelligibility of electrolaryngeal speakers. Steven R. Cox and Philip C. Doyle (Health and Rehabilitation Sci., Western Univ., Elborn College Rm. 2200, London, ON N6G1H1, Canada, scox47@uwo.ca)

Electrolaryngeal (EL) speech is often used by individuals who undergo total laryngectomy (Barney, Haworth, & Dunn, 1959; Doyle, 1994). Unfortunately, current EL devices continue to produce an unnatural and mechanical sound quality that can impact speech intelligibility (SI) (Doyle & Eadie, 2005; Hillman, Walsh, Wolf, Fisher, & Hong, 1998; Meltzner & Hillman, 2005). Clear speech (CS) requires speakers to produce speech "as clearly as possible" in an attempt to improve the overall understandability of speech (Picheny, Durlach, & Braida, 1985, 1986) in those with communication disorders (Beukelman, Fager, Ullman, Hanson, & Logemann, 2002; Tjaden, Sussman, & Wilding, 2014). This study addressed alterations in SI through the application of CS by EL speakers. Twelve naïve listeners orthographically transcribed CVC words produced by 10 EL speakers in both habitual and CS modes. Findings indicate that SI improved 1.5% for words spoken using CS (M = 53.25%, *range* = 47 to 64%) compared to words spoken using habitual speech (M = 51.75%, *range* = 41 to 62%). No significant differences in SI were found in either word-initial or word-final phonemes across speaking conditions. Clinical implications of these findings will be discussed.

3pSC3. Judgments of self-identified gay and heterosexual male speakers of American English: Certain phonemes are more salient than others in determining sexual orientation. Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Munson, McDonald, DeBoe, & White, 2006; Tracy, Bainter, & Satariano, 2015) has demonstrated that, upon hearing a relatively short utterance, listeners were able to differentiate between self-identified gay and heterosexual male speakers of American English. Furthermore, Tracy *et al.* (2015) discovered that listeners primarily relied on certain vowels (e.g., /æ/, /eɪ/, /ɛ/, /iː/, /oʊ/, /ɑ/, and /uː/) and consonants (e.g., /l/, /n/, and /s/) to determine a speaker's sexual orientation. The present study, based on data collected by Tracy (2015), investigated whether listeners relied on other phonemes when forming their sexual orientation judgments. Listeners were presented with utterances that either included one, two, or three consonants, or one, two, or three vowels. Upon hearing a single phone, such as /ɪ/, /^/, /ɹ/, /θ/, and /j/, listeners were able to differentiate between the speakers. With respect to the consonants, listeners' performance improved as the number of certain consonants (e.g., /l/, /n/, /ɹ/, /θ/, and /j/) in the utterance increased. For example, the utterance /ɹ, j, θ/ resulted in better performance than the utterance /f, m, v/. The results indicate that certain phonemes, more than others, can indicate a speaker's sexual orientation.

3pSC4. Are phonetic contrasts enhanced in clear speaking styles? Outi Tuomainen and Valerie Hazan (Speech, Hearing and Phonetic Sci., UCL, Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, o.tuomainen@ucl.ac.uk)

When asked to speak clearly, talkers make adaptations to various acoustic characteristics of their speech. Do these adaptations specifically enhance phonetic contrasts or just result in more global enhancements? For phonetic contrasts, increased discriminability could be achieved by increasing between-category distance, reducing within-category dispersion or both. The LUCID corpus contains 32 iterations per consonant for each of 40 adults for the /s/-/ʃ/ and /p/-/b/ contrasts. Iterations were obtained via picture elicitation in a sentence context in two conditions: when asked to speak casually and clearly. Friction centroids were measured for /s/-/ʃ/ and voice onset times for /p/-/b/. For /s/-/ʃ/, although there was significantly greater distance between centroids in the clear speech condition, within-category dispersion did not differ across speaking styles and there was no significant increase in overall discriminability in the clear condition. For /p/-/b/, in the clear condition, there was a significant increase in VOT distance, a significant decrease in within-category dispersion, resulting in an overall increase in discriminability. These results show that clear speaking styles can lead not only to global enhancements to speaking rate, intensity, pitch characteristics but also to increases in discriminability at the segmental level, although this may be contrast-specific.

3pSC5. Effects of speaking style and context on online word recognition in adverse listening conditions. Suzanne V. van der Feest (Linguist, The Univ. of Texas at Austin, 305 E 23rd St., Austin, TX 78712, suzanne@austin.utexas.edu), Samantha Moses, Amanda Clark (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), Danielle Parsons, and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX)

This study investigated the time-course of word recognition in adverse listening conditions. Specifically, we examined the effect of different listener-oriented speaking styles and semantic context on lexical access in quiet and in noise. Young adult listeners participated in an online visual word recognition experiment. They heard sentences with a high- versus low-predictability semantic context, produced in Conversational (CO), Infant-Directed (IDS), and Clear (CS) speech while fixating two pictures on a screen: a target that matched the last word of the auditory stimulus and a distractor. All sentences were presented either in quiet, or mixed with speech-shaped noise at a −5 dB SNR. Results showed that IDS provided similar perceptual benefits to adult listeners as CS. Relative to low-predictability CO baseline, IDS and CS increased speed of word recognition for high-predictability sentences, in quiet and in noise equally. However, in the quiet condition, lexical access was eventually facilitated by contextual cues even in CO, but listeners in noise reliably focused the target only when a combination of contextual cues and exaggerated acoustic-phonetic cues was available. These findings suggest that both semantic cues and listener-oriented acoustic enhancements are needed for reliable and rapid lexical access, especially in adverse conditions.

3pSC6. Spectral characteristics and formant bandwidths of English vowels produced by American males with different speaking styles. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

Speaking styles tend to have an influence on spectral characteristics of produced speech. There are not many studies on the spectral characteristics of speech because of complicated processing of too many spectral data. This study examined spectral characteristics and formant bandwidths of English vowels produced by nine American males with different speaking styles: clear or conversational styles; high- or low-pitched voices. PRAAT was used to collect pitch-corrected long-term averaged spectra and bandwidths of the first two formants of 11 vowels in the speaking styles. Results showed that the spectral characteristics of the vowels varied systematically according to the speaking styles. The clear speech showed higher spectral energy of the vowels than that of the conversational speech while the high-pitched voice did the same over the low-pitched voice. Second, there was no statistically significant difference between B1 and B2 in the speaking styles. B1 was generally lower than B2 reflecting the source spectrum and radiation effect. However, there was a statistically significant difference in B2 between the front and back vowel groups. The author concluded that spectral characteristics reflect speaking styles systematically while bandwidths measured at a few formant frequency points do not reveal style differences properly.

Plenary Session and Awards Ceremony

Christy K. Holland, Chair
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Ian C. Bruce – For contributions to models of auditory-nerve fibers

Michele B. Halvorsen – For contributions to the understanding of the effects of sound on fish

Valerie L. Hazan – For contributions to the understanding of the intelligibility of speech

Kirill V. Horoshenkov – For contributions to outdoor sound propagation, remote sensing, and acoustics of porous materials

Wilfried Kausel – For contributions to the understanding of the acoustics of brass instruments

Andone C. Lavery – For contributions to the understanding of ocean microstructure and biology

Jose Sanchez-Dehesa – For contributions to the theory and development of acoustic metamaterials

Introduction of Award Recipient

Carl Wunsch, recipient of the 2015 Walter Munk Award
for Distinguished Research in Oceanography Related to Sound and the Sea
a joint award of The Oceanography Society, the Office of Naval Research, and
the Office of the Oceanographer of the Navy

Presentation of Awards

Kelly Servick, recipient of the Science Writing Award in Acoustics for Journalists for her article "Eavesdropping on Echosystems" (*Science Magazine*, 21 February 2014)

Trevor Cox, recipient of the Science Writing Award for Professionals in Acoustics for his book *The Sound Book: The Science of the Sonic Wonders of the World* (W.W. Norton & Company, 2014)

Yang-Hann Kim, recipient of the 2015 Rossing Prize in Acoustics Education

Allan D. Pierce recipient of the Distinguished Service Citation

Paul D. Schomer recipient of the Distinguished Service Citation

John L. Butler, Silver Medal in Engineering Acoustics

Roy D. Patterson, Silver Medal in Psychological and Physiological Acoustics

Brian G. Ferguson, Silver Medal in Signal Processing in Acoustics

John J. Ohala, Silver Medal in Speech Communication,

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday, 3 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Orlando
Acoustical Oceanography	7:30 p.m.	River Terrace 2
Animal Bioacoustics	7:30 p.m.	City Terrace 9
Architectural Acoustics	7:30 p.m.	Grand Ballroom 3
Physical Acoustics	7:30 p.m.	St. Johns
Psychological and Physiological Acoustics	7:30 p.m.	Grand Ballroom 7
Structural Acoustics and Vibration	7:30 p.m.	Daytona

Committees meeting on Wednesday, 4 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Clearwater
Signal Processing in Acoustics	8:00 p.m.	City Terrace 7

Committees meeting on Thursday, 5 November

Committee	Start Time	Room
Musical Acoustics	7:30 p.m.	Grand Ballroom 1
Noise	7:30 p.m.	Grand Ballroom 2
Underwater Acoustics	7:30 p.m.	River Terrace 2

ACOUSTICAL SOCIETY OF AMERICA
DISTINGUISHED SERVICE CITATION

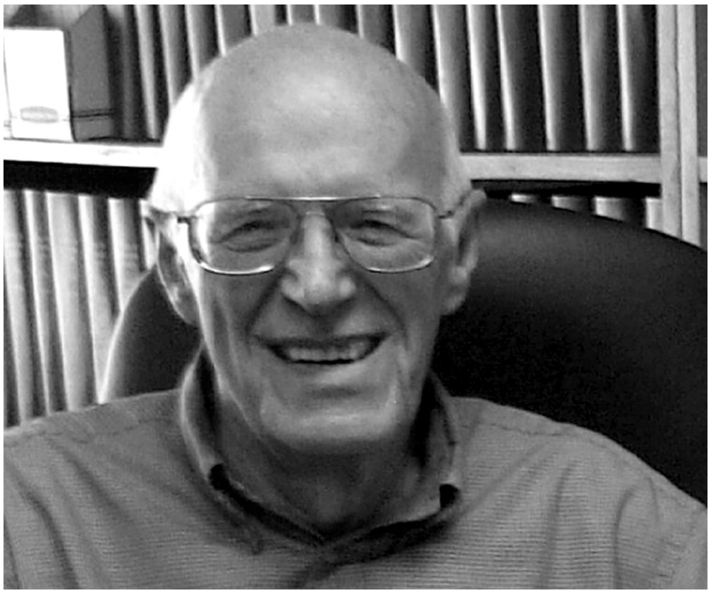
Allan D. Pierce

2015

The Distinguished Service Citation is awarded to a present or former member of the Society in recognition of outstanding service to the Society.

PREVIOUS RECIPIENTS

Laurence Batchelder	1972
Robert W. Young	1973
Betty H. Goodfriend	1973
Gerald J. Franz	1974
Robert T. Beyer	1978
Henning E. von Gierke	1978
R. Bruce Lindsay	1981
William S. Cramer	1984
Stanley L. Ehrlich	1986
Samuel F. Lybarger	1986
Frederick E. White	1987
Daniel W. Martin	1989
Murray Strasberg	1990
William J. Cavanaugh	1994
John C. Burgess	1996
Alice H. Suter	1997
Elaine Moran	1999
John V. Bouyoucos	2000
F. Avril Brenig	2000
Thomas D. Rossing	2006
Charles E. Schmid	2008
Uwe J. Hansen	2011
Richard Stern	2011



ENCOMIUM FOR ALLAN D. PIERCE

. . . for his excellent service to the Acoustical Society of America, and especially for his 15 years of service as Editor-in-Chief

JACKSONVILLE, FLORIDA • 4 NOVEMBER 2015

In their excellent article about Allan Pierce in the spring, 2015 issue of *Acoustics Today*, Jerry Ginsberg and Peter Rogers made one of their very rare scholarly mistakes. Specifically, they said: "If the Acoustical Society has any awards for senior members that Allan Pierce has not received, he would be the leading candidate…None are left…." Uncharacteristically for these two very exacting scholars, they missed the Distinguished Service Citation, which we are presenting to Allan Pierce today.

Writing an encomium that says something new about Allan Pierce at this point in time is actually quite a challenging task. Between Jerry and Peter's *Acoustics Today* article and Allan's 2005 Gold Medal encomium, Allan's academic career and personal traits have been rather well documented, at least in forms shorter than a full book. And there are other encomia for Allan's numerous awards, as well. So where does one begin?

Fortunately, I have had the privilege of knowing Allan as a close colleague and fellow journal editor for close to two decades now. So, hopefully, I have enough personal experience with Allan to add a few new stories to the collection.

I was introduced to Allan Pierce by the late Bill Carey, who was Allan's close friend and colleague at Boston University (BU), where Allan had put together a strong acoustics group. I knew Bill for years as a colleague doing shallow water acoustics, and when Allan hired Bill, I was drawn into Allan's orbit as well. My first impression of Allan, which hasn't changed much over the years, was: "where the heck does this guy get all his energy?!" Allan would travel from Cape Cod to Boston University at four or five in the morning, work on his class lecture notes and the Journal, teach his classes, and then settle down for bull sessions with colleagues at BU before heading home for the evening. Even given his well-known power naps, Allan displayed energy more in keeping with a "twenty-something" young faculty member than a senior professor.

Since this encomium is about Allan's service to the Acoustical Society of America (ASA), and particularly his fifteen years of service as ASA's Editor in Chief, let me continue mostly in that vein. When I succeeded Bill Carey as Editor-in-Chief of the IEEE *Journal of Oceanic Engineering* (IEEE JOE) in 1999, I became the junior member of the "Pierce/Carey Editor-in-Chief's Club," which was perhaps the best education as an apprentice that one could find in editing scientific and engineering journals. At that time, electronic publishing was just coming into its own, and Allan was leading the way with the *Journal of the Acoustical Society of America*. Allan also saw the value and promise of Bob Apfel's *Acoustics Research Letters Online* (ARLO), which combined rapid publication, multimedia, and open access all in one package. This was in 1999, well in advance of the curve of electronic publication and the "Open Access Revolution." Allan has never been shy about giving new ideas and new technologies a try, and is usually quite perceptive about which of these will prove robust over the years. When he saw how well the IEEE Oceans conference proceedings were doing, and how they allowed publication by practitioners and people unlikely to follow up their work with a full Journal article, he decided to inaugurate the *Proceedings of Meetings on Acoustics* (POMA), which has become increasingly popular over the years. He also was highly supportive of popular acoustics magazines, not just academic journals. *Acoustics Today*, first under Dick Stern and now with Art Popper, has enjoyed strong support from the ASA's Editor-in-Chief of publications. Allan liked to "get the word out" to people in a variety of ways.

In more recent years, Allan has shown his strong innovative and communicative touch as Editor. Regarding the current "Open Access Revolution," Allan has been quite supportive of it, despite the fact that it will modify our long-standing subscription-based business model. Allan championed *Acoustics Today*, *POMA*, and *JASA-EL* in open access format, and this has worked nicely. He also touted, at the end of his term as Editor, making Reviews, Tutorials, Special Invited Articles, and Forums open access, so as to engage a larger readership. This plan is now being implemented. And Allan also started the exploration by ASA to the larger world of publishing possibilities. Allan wanted to look at the entire publishing universe to make sure ASA was at the cutting edge. He initiated the Request for Proposals that led ASA to interview twenty different publishers, representing all facets of the publishing process. We now have a much broader worldview than before that exercise. Though the changes made are not so drastic (ASA has adopted Editorial Manager® as its peer-review package), the mindset of keeping up with the latest publishing trends has become a part of ASA's culture, and it is a welcome one.

However, Allan has still maintained the "old school" culture in one very important way: quality. Allan has very high technical standards, and moreover has a broad scientific background that allows him to understand and assess nearly every paper that he looks at. Given the diversity of topics that ASA's publications cover, that is not an easy task. And while Allan's primary criterion has always been the technical content of a paper or article, he is also fiendishly good on the stylistic elements of good writing. Independent of the native language of the author, Allan has always insisted that the overall writing style be clear and professional, a standard that we all highly applaud.

Allan has served on numerous administrative and technical committees within the Society (including Executive Council), has chaired and contributed to innumerable technical sessions, has written by far the best-selling book on acoustics (proceeds of which go to ASA), has mentored a number of students who have gone on to be leaders within the ASA, and bottom line, has been an amazing colleague to us all. Allan has been a giant intellectual and personal leader within the ASA, and that is perhaps his most important service.

To close, let me just let my esteemed colleagues Jerry Ginsberg and Peter Rogers know that, with the presentation of this Distinguished Service Citation, their reputation for accuracy has now been fully restored. And their last sentence of their article now has been made true on a Society level: "…the best award we can think of would be if you would thank him for his contributions when you see him." So, from the entire ASA, Thank You, Allan!

JAMES F. LYNCH

ACOUSTICAL SOCIETY OF AMERICA
DISTINGUISHED SERVICE CITATION

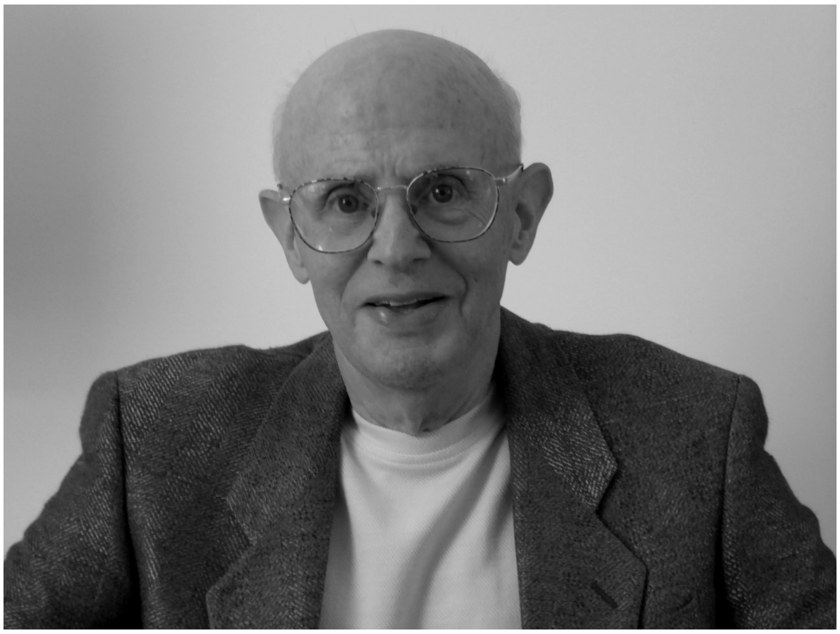
Paul D. Schomer

2015

The Distinguished Service Citation is awarded to a present or former member of the Society in recognition of outstanding service to the Society.

PREVIOUS RECIPIENTS

Laurence Batchelder	1972	Murray Strasberg	1990
Robert W. Young	1973	William J. Cavanaugh	1994
Betty H. Goodfriend	1973	John C. Burgess	1996
Gerald J. Franz	1974	Alice H. Suter	1997
Robert T. Beyer	1978	Elaine Moran	1999
Henning E. von Gierke	1978	John V. Bouyoucos	2000
R. Bruce Lindsay	1981	F. Avril Brenig	2000
William S. Cramer	1984	Thomas D. Rossing	2006
Stanley L. Ehrlich	1986	Charles E. Schmid	2008
Samuel F. Lybarger	1986	Uwe J. Hansen	2011
Frederick E. White	1987	Richard Stern	2011
Daniel W. Martin	1989		



ENCOMIUM FOR PAUL D. SCHOMER

. . . for excellent service as Standards Director of the Acoustical Society of America and for leadership in the development of standards in acoustics and the standards program

JACKSONVILLE, FLORIDA • 4 NOVEMBER 2015

Paul Schomer was born and raised in Chicago. By the time he was five, Paul knew he wanted to be an engineer. After watching a neighbor build a fence, he told his mother that the neighbor was "putting it in wrong."

Paul enrolled in the University of Illinois, earning a BS in 1965 in Electrical Engineering. He received an MS in Electrical Engineering-Acoustics from the University of California in 1966 and Ph.D. in Electrical Engineering-Acoustics from the University of Illinois at Urbana-Champaign in 1971. He met his wife Susan at a Jewish youth group in Chicago while she was in high school and he was a student at the U of I. They married in 1966 and have lived in Champaign since. They have two children, Beth and Jeffrey, and two grandchildren Madeline and Russell, and he is a terrific father and grandfather – enjoying introducing new activities to them.

After completing his education, Paul went to work for the U.S. Army Corps of Engineers in Champaign. He also became an adjunct professor in Electrical Engineering-Acoustics at the University of Illinois. About the same time, Paul joined the Illinois Task Force on Noise formed by the Illinois Institute for Environmental Quality to draft standards, which were used in noise regulation by the State of Illinois.

Paul joined the Acoustical Society of America (ASA) as a student in 1968 and became a regular attendee at ASA meetings. He soon became involved in the activities of the ASA standards committees beginning a long period of service as chair of a working group on blast noise in 1975. He was actively involved in ASA's standards committees S1 Acoustics, S3 Bioacoustics, and S12 Noise. Paul has published just under 60 research papers in peer-reviewed journals – the majority, about 30, in the *Journal of the Acoustical Society of America*. He was elected a Fellow of ASA in 1981 and is also active in other organizations, including the Institute of Noise Control Engineering (INCE), where he served in numerous leadership positions including Executive Director (2001-2003). He was elected a Fellow of INCE in 2010.

In 1983, Paul became chair of the working group on outdoor community noise. One of his major accomplishments was the leadership role he played in the development of the six-part S12.9 series of environmental noise standards.

Paul served in leadership positions in S12 beginning as Vice Chair (1991-96) and then Chair (1996-2002). In 1995 he became Chair of the U.S. Technical Advisory Groups to the International Organization for Standardization (ISO) standards committees on Acoustics and Noise. Since 1993, he has convened the ISO Working Group on environmental noise.

In 2002, upon the untimely death of Daniel Johnson, he became the Chair of the ASA Committee on Standards (ASACOS) and was appointed Standards Director by the Executive Council. Although Paul did not expect to become Standards Director when he did, he was well prepared for the job. That was fortunate since ASA was in the midst of one of the most hard-fought standards battles in its history: defense of the classroom acoustics standard [American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (ANSI/ASA S12.60)].

This groundbreaking standard was approved by ANSI Accredited Standards Committee S12 in 2001, and an appeal was filed immediately by the Air-Conditioning, Heating and Refrigeration Institute (AHRI). In addition, several other organizations appeared on the scene and petitioned the American National Standards Institute (ANSI) to withdraw the standard, claiming that it was not in the public interest. The appeal process was protracted. Paul was joined by then-President Dick Stern, testifying at the critical hearing in 2003 which resulted in a successful outcome.

Even after the appeal was rejected, Paul knew that the opponents needed to be engaged and not just ignored. He reached out to work with AHRI and the Modular Building Institute. Meetings were held with these important stakeholders, and Paul carefully listened to their concerns. He invited them to become constructive partners by co-chairing two new Working Groups with him. Together they led the effort to revise the first edition of ANSI S12.60 to its current two-part format: Part 1 focuses on permanent structures and Part 2 on special characteristics of relocatable classrooms. At the outset, there was suspicion among the parties but Paul has a remarkable ability to turn adversaries into allies. Paul's outreach to industry also resulted in sponsorships that have allowed ASA to make the two classroom acoustics standards available to the public at no cost. Paul is now focused on the goal of seeing the classroom acoustics standards referenced in building codes and other standards.

Many ASA members know that Paul Schomer retired from the post of Standards Director in 2015 after serving for 13 years. The position of Standards Director was created in ASA in 1978, and Paul is the seventh to serve. During his tenure, Paul was conscious of carrying on the tradition of former ASA Standards Directors and their predecessors who led the standards program before the title was established. These leaders of ASA Standards were also leaders in ASA and in the science of acoustics including Laurence Batchelder, Leo Beranek, Kenneth Eldred, Tony Embleton, William Galloway, Daniel Johnson, William Melnick, and Henning von Gierke.

Paul worked to raise the visibility of ASA's Standards Program within the Society and to increase the involvement of ASA members in development and use of ASA's standards. Implementation of the program to provide free standards as an ASA member benefit is just one example. Under his innovative leadership, ASACOS now sponsors and co-sponsors sessions at meetings in an effort to increase members' awareness of ASA's significant role as an ANSI-Accredited Standards Developing Organization.

Paul is known for his boundless energy and enthusiasm for translating acoustics research into practical applications for the public benefit. One of his important contributions is helping to bridge the gap between standards and public policy. Paul has been a tireless worker for implementation of noise standards in buildings, classrooms, hospitals, wilderness areas, soundscapes, and now, wind-turbine farms. He takes every opportunity to share his love for acoustics and engineering with future generations of scientists. When his children were in high school, Paul started a Boy Scout Explorer Engineering Group encouraging high school students with engineering projects. To this day, Paul hires and trains high school students to assist in his consulting, giving them research assignments and having them co-author papers for presentation at ASA meetings and publication in *JASA*.

Paul's dedication to ASA, exemplified by his leadership in the ASA Standards Program, has surely earned him this Distinguished Service Citation.

BRIGITTE SCHULTE-FORTKAMP
GILLES A. DAIGLE
PEGGY B. NELSON
ROBERT D. HELLWEG, JR.
SUSAN BLAESER

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Engineering Acoustics

John L. Butler

2015

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Harry F. Olson	1974	Alan Powell	1992
Hugh S. Knowles	1976	James E. West	1995
Benjamin Bauer	1978	Richard H. Lyon	1998
Per Vilhelm Briiel	1982	Ilene J. Busch-Vishniac	2001
Vincent Salmon	1984	John V. Bouyoucos	2004
Albert G. Bodine	1986	Allan J. Zuckerwar	2007
Joshua E. Greenspon	1989	Gary W. Elko	2011

ENCOMIUM FOR JOHN L BUTLER

. . . for advancing the field of acoustic transducer and transducer array design

JACKSONVILLE, FLORIDA • 4 NOVEMBER 2015

John L. Butler, better known to his friends as Jack, has been interested in electroacoustic transducers, how they work, and how to improve them, for most of his career. Unlike those of us who have occasionally dabbled in this field, Jack has devoted his professional life to transduction theory and design ranging from original loudspeaker concepts to innovative underwater transducer designs based on the superposition of acoustical modes. His consulting firm, Image Acoustics, was founded in 1974 and has been operated by John and his family ever since then. His children have been sufficiently inspired by his example to enter the field themselves as engineers specializing in transducer design and development. Two of his children, Alex and Tori, continue to work with their father at his company. Another son, Stephen, is the senior transducer designer and builder at the Naval Undersea Warfare Center in Newport, Rhode Island.

Jack Butler received a Ph.D. in electrical engineering from Northeastern University after earning the Sc. M. degree in physics from Brown University. He has been a driving force in educating both the transducer design community and the sonar systems engineering community for decades and his reputation as an academic and innovative engineer is renowned.

Jack has amassed a broad background in the practical aspects of transducer design through his experience with various types of loudspeakers, underwater projectors, hydrophones, and transducer materials. He has been instrumental in the development of the theory needed to understand transducers and the computer models to aid in their design. Jack's most lasting contribution to students and practitioners of engineering acoustics is his co-authorship with Charles Sherman of *Transducers and Arrays for Underwater Sound* in 2007, a comprehensive textbook that compiles much of the experience of the authors' long collaboration in transducer and array problems. Such an authoritative, comprehensive textbook has long been needed in the transducer design field. This text is becoming the accepted teaching means for future underwater transducer designers and academics with a second edition in the works.

Jack has published over 34 peer-reviewed papers, almost exclusively in the *Journal of the Acoustical Society of America* (JASA) and has received 27 patents involving transducer design. He was elected a Fellow of the Society in 1977. He continues to educate with rigorous analyses of issues of relevance such as heat generation in active projectors and methods for evaluation of newer transducer materials. His 2011 paper in JASA entitled "The modal projector" initiated a series of innovative transducer designs that exploit the superposition of acoustical modes radiated from a transducer.

Jack's 'style' is to approach a problem presented to him as a challenge. He thinks through the physics of the issues and comes up with an analytical model of the transducer he has conceived or the array configuration imagined. He then often writes a computer program that permits him to analyze the various iterations that can then be tested for accuracy and validity. A good example is the program FLEXT that he developed for the synthesis and analysis of flextensional transducers. By inclusion of the mathematics for interactions between transducer elements, it has become the program of choice for designing flextensional transducer arrays of several different classes. This program has replaced the traditional point source array model as the most realistic way to understand complex array interactions and behavior. In the past ten years or so he has supplemented this synthesis approach to transducer design with finite element analysis so as to more fully understand transducer behavior, but with the continued emphasis on the need for understanding rather than simply knowledge.

Jack does not limit himself to underwater acoustics. He has designed and manufactured audio loudspeakers for years, both as a business and a personal interest. One recent effort incorporates a unique ribbon tweeter design with the upper cross-over frequency chosen to minimize potential distortion in the primary listening band. He has also consulted in acoustic design in general and is an outstanding critic of all things acoustic.

In perusing the list of Silver Medal honorees in Engineering Acoustics, one is naturally drawn to the name of Harry Olson who received this award in 1974. Like Olson, Jack Butler continues the tradition of pioneers of acoustical engineering who have innovated, designed, built, taught, and served as a role models for our profession.

JAN LINDBERG
MARK MOFFETT

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Psychological and Physiological Acoustics

Roy D. Patterson

2015

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Lloyd A. Jeffress	1977	Neal F. Viemeister	2001
Ernest Glen Wever	1981	Brian C. J. Moore	2002
Eberhard Zwicker	1987	H. Steven Colburn	2004
David M. Green	1990	William A. Yost	2006
Nathaniel I. Durlach	1994		

von Békésy Medal

The von Békésy Medal is presented to an individual, irrespective of nationality, age, or society affiliation, who has made an outstanding contribution to the science of psychological and physiological acoustics, as evidenced by publication of research results in professional journals or by other accomplishments in the field.

PREVIOUS RECIPIENTS

Jozef J. Zwislocki	1985
Peter Dallos	1995
Murray B. Sachs	1998
William S. Rhode	2010
M. Charles Liberman	2012

ENCOMIUM FOR ROY D. PATTERSON

. . . for contributions to understanding pitch and timbre perception, and for computational modeling of auditory representations

JACKSONVILLE, FLORIDA • 4 NOVEMBER 2015

Roy D. Patterson is one of the most productive and creative auditory scientists of his generation, working in an era of critical and breath-taking advances in Psychological and Physiological Acoustics. His work spans nearly five decades, three countries, and several basic and applied research areas within the field. Roy was born in Boston but he grew up in Toronto and studied at the University of Toronto, first in Chemical Engineering and then in Experimental Psychology. He continued his education at the University of California at San Diego (UCSD), earning a Ph.D. in 1971 in Experimental Psychology in Dave Green's lab. From the start, Roy's concern was to develop a computational model of auditory perception, a goal that he later achieved with great success. In Green's lab, he began with research on pitch perception, the topic of his dissertation, and auditory masking, where he extended Fletcher's "critical band" concept of frequency analysis with a continuous, frequency-weighting function referred to as the "auditory filter." The results of the two research streams were then used to construct a functional model of the peripheral processing implied by auditory perception.

Roy and his wife, Karalyn Patterson, a world-renowned cognitive neuroscientist, moved to Canada after receiving their Ph.D. degrees from UCSD. Roy worked for the Defence and Civil Institute of Environmental Medicine, where he used his model of auditory masking to improve communications and auditory warnings in helicopters and Canada's new long-range patrol aircraft, the Aurora.

In 1975 Karalyn and Roy moved to Cambridge, England, where they both took positions at the Applied Psychology Unit (APU) of the Medical Research Council (MRC). Roy's research on auditory perception led to the design of warning systems for a range of noisy environments at the request of the Royal Airforce, the Civil Aviation Authority, British Rail, the London Fire Brigade and the International Organization for Standardization (ISO) committee on hospital warnings. Soon a key ingredient of Roy's career emerged, i.e., fruitful collaboration with researchers from all over the world. There were extended research visits with collaborators in both directions, and many of the collaborators' students and post-docs then held post-doctoral positions in Roy's lab. During this time, Roy continued his studies of pitch which eventually led to his "pulse-ribbon" model of pitch and phase perception. Given Roy's deep appreciation of music, it is not surprising that a major part of his life's work has been to understand pitch perception. In Cambridge itself, Roy's main collaborator was Brian Moore in the Psychology Department. Together, they and their research groups continued to develop the "notched-noise" method for measuring auditory frequency selectivity, eventually producing the gold standard technique for such measurements. At that time, the selectivity was summarized with a rounded-exponential, or "roex," auditory filter.

In 1990, Roy's group developed a dynamic Auditory Image Model (AIM) to investigate the internal representation of natural sounds like speech and music, and the role of neural phase-locking in auditory perception. At the time, the computational load of such models was dominated by the convolution required to simulate cochlear filtering. To make AIM workable, they developed a particularly efficient version of the gammatone impulse response, and the resultant Gammatone Auditory Filterbank has arguably become the most widely used method for simulating auditory spectral analysis at moderate sounds levels. At high sound levels, the magnitude spectrum of the auditory filter becomes progressively more asymmetric which means that the carrier of the impulse response must chirp to some degree. Several research groups proposed forms of Gammachirp Auditory Filter which led a post-doc in Roy's lab (Toshio Irino) to consider what the optimum form would be in the sense of providing optimum time-frequency trading when the frequency dimension is logarithmic. In a series of papers, physiological and perceptual data were used to tune the parameters of this new gammachirp filter system, and the resultant "dynamic, compressive, Gammachirp Auditory Filterbank" is now rapidly replacing the Gammatone Filterbank in situations where accuracy at higher stimulus levels is required.

In 1997, Roy moved from the APU to the Physiology Department of the University of Cambridge, where he set up the Centre for the Neural Basis of Hearing (CNBH) with the physiologist Ian Winter and two external collaborators: Ray Meddis at Essex University who was developing physiologically informed models of different auditory cell types, and Tim Griffiths at Newcastle University Medical School, who was developing fMRI techniques to locate the neural activity associated with pitch processing in the auditory pathway. Toshio Irino became Roy's main international collaborator. At the heart of the derivation of the optimum gammachirp filter is a form of scale transform which Roy and Toshio realized might explain why auditory perception is so robust to variation in source size. (Humans understand speakers no matter what their size and know which of two speakers is larger no matter what they are saying.) This led Roy and Toshio to develop a scale-shift invariant version of the auditory image where, for example, the vowels of children and adults have the same form. They performed perceptual experiments demonstrating just how robust the perception of speech and music is to variation in acoustic scale, and they set out the implications for "hearing machines" intended to process the sounds of speech and music. Under Roy's leadership, the CNBH established a worldwide network of psychoacousticians, auditory physiologists, neuro-imaging specialists, and computational modelers who collaborated to produce hundreds of scientific papers on auditory processing and perception, using a wide range of techniques (behavioral experiments with humans and other animals, single-unit physiology, and increasingly, brain imaging experiments to associate auditory processing with specific neural regions in the auditory pathway).

No review of Roy's endeavors would be complete without a mention of his hobbies: gardening, gourmet food and fine wine. Roy grows exotic conifers (giant Sequoias and their relatives from around the world). And, without doubt, the highlight of any visit to Cambridge is an invitation to dinner at the Pattersons'. Dinners typically include Cambridge colleagues, so not only are the food and wine outstanding, the conversation is stimulating.

The past fifty years has seen an explosion of new knowledge about auditory perception. Much of that knowledge can be traced, both directly and indirectly, to the contributions of Roy D. Patterson – a most deserving basis for the award of the Silver Medal in Psychological and Physiological Acoustics.

WILLIAM A. YOST
MARJORIE R. LEEK
RAYMOND MEDDIS

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Signal Processing in Acoustics

Brian G. Ferguson

2015

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Edmund J. Sullivan	2010
Theodore G. Birdsall	2011

ENCOMIUM FOR BRIAN G. FERGUSON

. . . for contributions to in-air and in-water acoustic classification, localization, and tracking

JACKSONVILLE, FLORIDA • 4 NOVEMBER 2015

Brian G. Ferguson began his research career in atmospheric physics, but eventually found his true calling in the field of signal processing.

Brian graduated from the University of Sydney (Australia) in 1972 with a B.Sc.(Hons) degree in Physics, together with both Pure and Applied Mathematics. He graduated with a Diploma of Education from the University of Sydney in 1973. He was awarded an M.Sc. in 1975 and a Ph.D. in 1982 by the University of New South Wales (Australia).

Brian Ferguson began his professional career as a physicist with the Australian Government's Department of Science in 1974 where he conducted research in solar terrestrial relationships. This encompassed the disciplines of solar physics, solar observation (optical and radio astronomy), ionospheric physics, and high-frequency radio wave communications. He began his signal processing career in 1984, when he joined the Submarine Sonar Group of the Royal Australian Navy Research Laboratory as a research scientist. Since then, he has continuously provided innovative and creative solutions to real-world problems involving towed and hull-mounted arrays, firearm projectile tracking, source localization, and aircraft tracking from undersea platforms, a problem that crosses the air-water interface. He has also made a number of significant contributions in the area of time delay estimation as evidenced by his chapter in the text *Coherence and Time Delay Estimation* (IEEE Press, 1993) considered by many to be the 'bible' in this area. Essentially all of this work has led, in some form, to inclusion in present day systems.

There are three characteristics associated with Brian Ferguson's work. It always embodies a novel approach, it is always experimentally verified, and it usually finds its way into real systems. There are few researchers who can make this claim. In Brian's case it has resulted in his recognition by the Australian Government as well as internationally. He is one of only three staff members elevated to the level of Principal Scientist in Australia's Defence Science and Technology Organisation (DSTO), an organization of 2300 people. It is Australia's second largest public funded R&D organization with an annual budget of $408 million. The selection panel recommended that Brian be appointed as Principal Scientist–Acoustic System Science because "he had a distinguished international reputation for his science and he provided leadership in a scientific discipline of importance to the DSTO. His program of work is highly relevant to DSTO's client requirements and the results he produces contribute to successful client outcomes through substantial personal contributions in his scientific field. In doing this, he contributes directly to DSTO's reputation for scientific excellence through the publication of new knowledge of high quality and development of external collaborative links."

Brian's more recent research has been focused on the localization of hostile mortar fire and sniper fire. According to the chief of the Acoustics and Electromagnetics Sensing Branch: "The US Army Research Laboratory (ARL) has a collaborative program that directly involves Dr. Brian Ferguson in the research and development of acoustic localization systems for pinpointing (1) hostile mortar fire and (2) enemy sniper fire. His extraordinary accomplishment in this area has proved to have the most laudable of impacts in that it saved the lives of coalition soldiers and innocent civilians during Operation Iraqi Freedom. This extraordinary outcome was the result of the deployment of an acoustic system that accurately geolocated mortar fire from insurgents, even in urban environments. The system was based on the Ferguson *et al.* seminal paper "Locating far-field impulsive sound sources in air by triangulation," published in the *Journal of the Acoustical Society of America* in 2002.

Brian is Australia's national leader in battlespace acoustics under The Technical Cooperation Program between the Governments of Australia, Canada, New Zealand, the United

Kingdom, and the United States. On the international scene, he is an invited member of NATO's Sensors and Electronics Technology Panel, and was a recipient of the NATO Research and Technology Organization Scientific Achievement Award in 2009, notwithstanding the fact that Australia is not a member of NATO.

Although it is not easy to travel to the United States from Australia, Brian does attend many meetings of the Acoustical Society of America (ASA). He has been a member of the Signal Processing Technical Committee since 2000 and was elected Fellow of the ASA in 1999. He has contributed to several of the Signal Processing Technical Committee's international student challenge problems, has chaired several special sessions at meetings, has conducted many reviews of manuscripts for the *Journal of the Acoustical Society of America*, and was awarded the first prize for his contribution to the Gallery of Acoustics.

Brian Ferguson is, by any measure, an asset to the Acoustical Society of America and the Australian defense science and technology enterprise. We are pleased to congratulate him for being awarded the Silver Medal in Signal Processing in Acoustics.

EDMUND J. SULLIVAN
JAMES V. CANDY

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Speech Communication

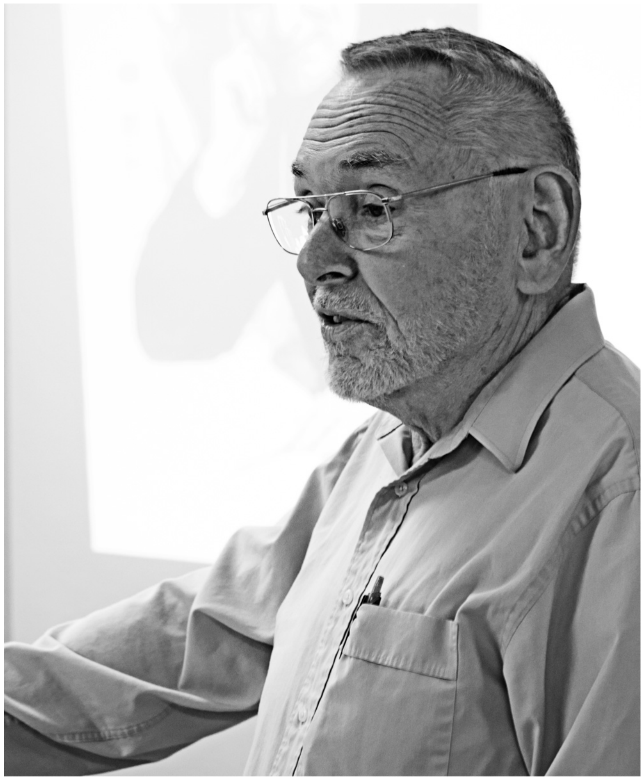
John J. Ohala

2015

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Franklin S. Cooper	1975	Patricia K. Kuhl	1997
Gunnar Fant	1980	Katherine S. Harris	2005
Kenneth N. Stevens	1983	Ingo R. Titze	2007
Dennis H. Klatt	1987	Winifred Strange	2008
Arthur S. House	1991	David B. Pisoni	2010
Peter Ladefoged	1994	Sheila E. Blumstein	2014



ENCOMIUM FOR JOHN J. OHALA

. . . for advancing the understanding of speech production and perception, and applying phonetic principles to the study of spoken language change over time

JACKSONVILLE, FLORIDA • 4 NOVEMBER 2015

John J. Ohala, emeritus Professor of Linguistics at the University of California, Berkeley and Research Scientist at the International Computer Science Institute, Berkeley, is a towering figure in the field of linguistic phonetics and its application to understanding the phonological structures of the world's spoken languages and the way they change over time. John was born in Chicago. He obtained a degree in English (from Notre Dame) before converting to linguistics at the University of California, Los Angeles (UCLA) where his Ph.D. was directed by Peter Ladefoged. At UCLA he met and married Manjari Agrawal (Ohala), herself a noted phonetician. His long career at Berkeley began in 1970.

John's contributions are richly varied. Many regard him as the founder of the field of laboratory phonology. A longstanding approach to phonology had been to sharply differentiate speakers' and listeners' cognitive representations of sound structures from the physical principles of phonetics. John's research agenda challenged this concept of the autonomy of phonology and phonetics. Through a masterful series of highly influential papers spanning from the 1970s through today, he demonstrated that neither is independent of the other. Moreover, he insisted on an experimental approach, arguing that, regardless of the relation between phonology and phonetics, rigorous testing of hypotheses was required whether they deal with the laws governing those systems or with humans' knowledge of sound systems. The 1986 volume titled *Experimental Phonology* that he edited with his former student Jeri Jaeger, and the special issue of the journal *Phonology* that he edited in the same year, were milestones in this (then) new approach to phonological investigation.

John also revolutionized the way that linguists think about historical sound change by emphasizing two important perspectives that had tended to be ignored. First, John insisted that perceptual processes play an equally crucial role as do articulatory ones in fostering sound change. This view is epitomized in his seminal 1981 paper "The listener as a source of sound change" (Chicago Linguistic Society). In accompanying detailed work he has pointed to specific ways in which the acoustic patterns of particular sounds are likely to bias perception and foster such a reinterpretation of the target, as in the case of labialized velars being interpreted as simple bilabial sounds. The second perspective is that of variation — not the obvious sort of variation that is readily apparent but what he called "hidden" variation in his equally influential 1989 paper "Sound change is drawn from a pool of synchronic variation." (in *Language Change: Contributions to the Study of its Causes,* ed. Breivik & Jahr). This variation comes (in part) from the imprecision of the control mechanisms in speech production, but is capable of generating just the kind of potential ambiguity that a listener might seize on in (mis-)interpreting the signal. John's contribution lies not only in framing and forcefully arguing for the novel viewpoint but in constructing and carrying out clever experiments, often in collaboration with his students at Berkeley, that test predictions arising from thinking along these lines.

An additional theme of paramount importance in his work is an enduring effort to examine the full extent of the ways that phonological patterns are shaped by the physical constraints governing the speech production system. In John's view, a large proportion of phonological patterning has a direct source in such constraints. An extremely influential statement of this view is in the 1983 paper "The origin of sound patterns in vocal tract constraints" (in *The Production of Speech,* ed. MacNeilage) and this is followed up in numerous other publications such as 2005's "Phonetic explanations for sound patterns." (in the festschrift for John Laver). One particularly insightful contribution in this vein (see, e.g., Ohala and Riordan 1979, "Passive vocal tract enlargement during voiced stops." In *Speech Communication Papers,* ed. Wolf & Klatt) has become known as the 'aerodynamic voicing constraint,' which among other things, clearly accounts for why there is a greater likelihood that /g/ rather than stops at more forward places of articulation is missing from a series of voiced stops — there is simply more volume and, more importantly, more compliant surfaces in the oral cavity to accommodate trans-glottal air flow in the latter situation, and this is crucial to sustain voicing.

John has also had another major impact on the field of speech communication through rehabilitating the concept that some part of the relationship between sound and meaning is non-arbitrary. In particular, he pointed out (following insights of biologist Eugene Morton) that not only animals but also humans are sensitive to the physical correlation between the size of an object and the frequency of the acoustic signal it is likely to generate. Hence, larger size — and derivative factors such as greater threat — is signaled by lower frequencies, and the converse by higher frequencies. Both the fundamental frequency of the voice source and the spectral shaping of the filtered sound output serve as cues. He argued that this 'frequency code' is implicated, for example, in the common occurrence of high front vowels in diminutives, as well as in predominant cross-linguistic patterns in intonation (1994's "The frequency code underlies the sound symbolic use of voice pitch" in *Sound Symbolism*, ed. Hinton, Nichols, and Ohala).

Not only has John been of major importance for his intellectual contributions to the field but he has served in many generous ways to develop the capacities of others and to give back to the community. He was the President of the International Phonetic Association from 1995-99, he organized the 2nd International Conference on Spoken Language Processing (ICSLP) in 1991 as well as the 14th International Congress of the Phonetic Sciences in 1999, he has been on the editorial board of *Phonetica* and other publications and served on many committees, juries and panels, and often has been a mentor to students and younger scholars not directly under his charge. In short he has given unstintingly of his time, wisdom and knowledge, as well as of his inexhaustible collection of jokes, to colleagues both senior and junior and to the academic community at large. His contributions have been recognized through an honorary doctorate (University of Copenhagen), and election as a fellow of several societies (Acoustical Society of America, Linguistic Society of America, American Association for the Advancement of Science, International Speech Communication Association). Along the way, when the questions that he wanted to address were in advance of the available technology (in his case, not an infrequent occurrence), he donned the hat of a speech engineer and developed his own instrumentation for taking aerodynamic, respiratory, laryngeal, and articulatory measurements.

In short, John Ohala is among the top few of the world's foremost scholars in phonetics and phonology. For the past 40 years, his pioneering, sometimes controversial, but consistently insightful work has opened up new lines of investigation into sound structure and into human knowledge of these structures, setting the trajectory for the field and inspiring—by his example—other researchers to be imaginative and to broadly embrace other disciplinary approaches in their investigations.

IAN MADDIESON
PATRICE SPEETER BEDDOR

Session 4aAA

Architectural Acoustics and Noise: Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance 1

Kenneth P. Roy, Cochair
Building Products Technology Lab, Armstrong World Industries, 2500 Columbia Ave, Lancaster, PA 17603

Donna A. Ellis, Cochair
The Division of Architecture and Engineering, The Social Security Administration, 415 Riggs Ave., Severna Park, MD 21146

Chair's Introduction—8:55

Invited Papers

9:00

4aAA1. Contributions to acoustical comfort in the built environment, 1956–2015. Jerry Christoff, Jim Good, John LoVerde, David W. Dong, and Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

When the senior author joined Paul S. Veneklasen & Associates in 1956, research was under way on speech intelligibility relative to background noise, not in buildings but in spacecraft! Back on earth, additional studies countered the then-popular idea that gypsum board construction was low quality with poor sound isolation, leading to the development of multi-layer gypsum board wall designs for office buildings and studios. Further investigations predicted speech intelligibility based on design features and led to development of prototype masking noise systems. A set of acoustical design elements (such as tall furniture barriers, sound-absorptive barriers and ceilings, and orienting workstations to maximize distance) emerged naturally from these studies. Within the last couple of decades, however, open-plan office design has evolved away from traditional design, incorporating fewer of these design elements. This provides opportunity to evaluate the efficacy of standard design elements in modern collaborative work areas. Occupancy surveys and acoustical measurements were performed in open office areas of six companies. The results verify the effectiveness of traditional acoustical design elements, but also indicate where typical acoustical design practices may be modified while maintaining acceptable acoustical performance in collaborative open office areas.

9:20

4aAA2. Capturing office acoustics in the design and construction process. Roman Wowk and Chris Papadimos (Papadimos Group, 222 Vallejo St., 4th Fl., San Francisco, CA 94111, roman@papadimosgroup.com)

Given the realities of the design and construction process, basic qualities for office acoustics including speech privacy, freedom from distraction and speech intelligibility are often compromised or neglected altogether. A scan of recent news articles not only yields numerous accounts of poor conditions, but several academic studies have also shown how poorly acoustics compares with other indoor environmental qualities. This cannot be overlooked as a factor affecting productivity and performance. While the marketplace has no shortage of potential solutions for office acoustics, no product or design upgrade alone is a substitute for good design and planning, regardless of its role in previous solutions. Several essential but independent elements must be considered together for successful and cost-effective outcomes. This necessitates an integrated approach with consensus between user expectations and any design limitations, early consideration of acoustics during space planning, and the right balance of upgrades to meet project criteria. However, even basic oversights during construction can unravel even a wholesome design without coordination between trades, verification of acoustic performance, and field review. This presentation will show through case studies how such a process has either been successfully executed or systematically neglected and how that has affected the outcome on various projects.

9:40

4aAA3. Architecture and the environment—How much do we care? Kenneth P. Roy (Bldg. Products Innovation Lab., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

We have been hearing about "green buildings" and the drive for energy efficiency and sustainability for quite some time now. And if we think about this, we will probably realize that this focus is really on the buildings and the cost to society that is incurred by these building in terms of environmental factors such as materials, air, and water. This of course does not address the primary reason to design, build, and operate buildings—which is really about "people." People are exposed to the indoor environmental quality, and up until now, acoustic design and performance have not been exactly exemplary, as shown by POE surveys such as by the CBE at Berkeley. So, what needs to be done to make this work??—Focus on the mission of the building, after all, the building is just a tool to help us do what we need to do. So, how do we address occupant satisfaction and performance with architecture?? And how do we integrate acoustic design into the emerging architectural trends?? That's what we need to discuss.

10:00–10:15 Break

10:15

4aAA4. Research methods to investigate the effects of acoustics on occupant comfort and productivity in the built environment.
Michelle C. Vigeant and William P. Bahnfleth (Graduate Program in Acoust. and Dept. of Architectural Eng., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu)

A number of factors affect building indoor environmental quality (IEQ), principally thermal environment (temperature, humidity, and air movement), indoor air quality (IAQ), lighting, and acoustics. Most studies on comfort or preference and virtually all dealing with health and productivity focus on thermal environment and IAQ. However, a recent meta-analysis showed that thermal comfort ranks only slightly higher in importance than acoustic comfort and IAQ (Frontczak & Wargocki 2011). Although not focused on acoustics, these prior studies do provide insight into appropriate methodologies. Perceived IEQ is the most widely studied since it is relatively simple to measure and can be compared to perception-based acceptability criteria. Some investigations go beyond formal surveys by including quantitative measures, such as overall productivity, student learning, and health. These data can be used for monetization of benefits and cost, where the typical finding in studies focused on the thermal environment and IAQ is as high as 10:1. Overall, the field of IEQ needs to move from perceived quality metrics to performance metrics, but more work is needed to establish sufficiently accurate quantitative measures. These advances will be especially beneficial to building occupants by providing better working environments both in terms of health and productivity.

10:35

4aAA5. The national opera theatre in Bucharest—Update of the room-acoustical properties.
Wolfgang Ahnert, Tobias Behrens (ADA Acoust. & Media Consultants GmbH, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu), and Radu Pana (Univ. of Architecture and Urbanism “Ion Mincu", Bucarest, Romania)

After the destruction of the old opera house in WWII, the new edifice was designed in 1952–1953. The hall and the pit was almost not changed over the time, so based on the original design the hall acoustics was like in classic opera house not very lively, i.e., very dry. Also, the mutual hearing of the musicians in the pit and between the singers on stage and the musicians was not good. The facelift of the hall involved significant changes of the orchestra pit. A new lift has been built, and the cover of the musicians in the pit has been reduced. Also, the secondary structure of all wall and ceiling parts in the pit has been modified. The same has been done in hall, the carpet in the stalls, and the galleries has been substituted by a parquet floor, the absorbing materials in the boxes, and other wall parts was exchanged by wood panels and the completely absorbing chairs have been changed to less absorbing ones. Acoustic measurements have been done before and after the refurbishment. The paper will describe all changes and the achieved results.

10:55

4aAA6. Source qualification environments and their impact on speech privacy measurements in open offices.
Sean Browne and Kenneth Good (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604, sdbrowne@armstrong.com)

The ASTM standards for measuring speech privacy in an open office call for specific loudspeaker dimensions and performance attributes. The standards also details specific testing environments required to qualify those attributes and set measurement reference levels. Often times, these environments are not accessible or practical during the course of a field measurement. Several additional test environments were explored in addition to the preferred and alternate test environments defined in ASTM E1179. The reference levels from these alternate environments were used, and the resulting speech privacy measurements were compared.

Session 4aAB

Animal Bioacoustics: Avian Bioacoustics

Micheal L. Dent, Cochair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Robert Dooling, Cochair
Psychology, Univ of Maryland, Baltimore Ave, College Park, MD 20742

Invited Papers

8:30

4aAB1. Auditory temporal resolving power in birds. Robert Dooling (Psych., Univ. of Maryland, Baltimore Ave., College Park, MD 20742, rdooling@umd.edu)

There has long been the notion that bird song contains acoustic features, which are inaccessible to human listeners. Field and laboratory studies suggest that birds hear both the spectral and gross temporal features of bird song much like we do. But how precisely birds perceive their auditory world still remains somewhat of a mystery. Here, I review psychophysical studies that provide evidence that birds have an exquisite sensitivity to temporal fine structure in their vocalizations putting some aspects of bird vocalizations out of reach of human hearing. Work with zebra finches, in particular, shows that these birds are capable of discriminating between positive and negative Schroeder harmonic complexes at high fundamental frequencies. These birds can also discriminate temporal fine structure changes in their vocal signals that are well below human thresholds. Whether this ability is used for communication is still not clear but it could underlie the better-than-expected ability of birds to localize sound and to assess the effects sound degradation and reverberation as a complex sound travels through the natural environment.

8:55

4aAB2. Seemingly simple songs: Black-capped chickadee song revisited. Christopher B. Sturdy (Psych., Univ. of AB, P-217 Biological Sci. Bldg., Edmonton, AB T6G2E9, Canada, csturdy@ualberta.ca)

Our lab has been studying songbird communication from an integrative perspective for over 13 years. A significant part of this program involves conducting bioacoustic analyses of vocalizations that are critical to survival. Initially, black-capped chickadees and their *chick-a-dee*call were our main research focus. In more recent years, we have turned our attention back to the *fee-bee* song, intensely studied by Weisman, Ratcliffe, and colleagues. Our re-examination of the *fee-bee* song revealed several, previously unreported features of this seemingly acoustically simple vocalization. We showed that songs contain regionalized cues for dominance status, and that females respond differentially to dominant song playback irrespective of their geographic origin. Moreover, females themselves produce a *fee-bee* song not previously reported in the literature. Female song is acoustically distinct from male song, and we have used operant conditioning experiments to identify features that can be used to identify sex of the singer. Finally, we have begun to explore neural response to different vocalizations, including song, and found that responses vary by singer, listener, and vocalization type. Our current efforts are aimed at unraveling the acoustic basis of communication in this nearly ubiquitous North American species.

9:20

4aAB3. How birds perceive the auditory scene. Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

How are birds able to effectively communicate for survival? How do they separate auditory objects that overlap on the tympanum? How do they perceive distance, overlapping sounds, and noisy signals? In many ways, birds decipher the auditory world in the same way humans do. We know they use intensity as a primary cue for distance perception, that they spatially segregate signals and noise to better hear certain signals, that they are susceptible to auditory illusions, and that spectrotemporally complex and biologically relevant signals are often better isolated than simple tonal sounds. We know these things and more from psychophysical studies in the laboratory where birds are trained to be reliable observers, using operant conditioning techniques in a controlled environment. Subjects are trained to peck keys for food reinforcement when they detect, discriminate, identify, or localize sounds. Years of research on the foundations of hearing in these animals leads us to the present day, where birds are asked much more complicated questions in studies that more closely resemble their natural auditory world. For birds, being able to hear and isolate communication signals in the environment is crucial, and we have discovered much about how they do this in complex acoustic environments.

9:45

4aAB4. Evolution of song complexity in Bengalese finches: Sexual selection and domestication as two factors. Kazuo Okanoya (Life Sci., The Univ. of Tokyo, 3-8-1 Komaba, Meguro-ku 153-8902, Japan, cokanoya@mail.ecc.u-tokyo.ac.jp)

Bengalese finches (BFs) are domesticated strains of wild white-rumped munias (WRMs) imported from China to Japan 250 years ago. BF songs are composed of multiple chunks and each chunk is a combination of 2–4 song notes. Chunks are then arranged in a finite-state probabilistic automaton. We studied how and why BFs sing such complex songs. We found the following facts. (1) The ancestral strain sing simpler songs. (2) There is high learning specificity in WRMs but not in BFs. (3) BFs have larger song control nuclei and higher level of glutamate receptor gene expressions than WRMs. (4) Both BF and WRM females prefer complex songs as measured by the nest string assay and males with complex songs are physically fitted than the males with simpler songs. These results promoted sexual selection scenario of song complexity in BFs. We further examined factors related with domestication. We examined songs of WRMs in subpopulations of Taiwan. Where there is a sympatric species to WRMs, songs were simpler. This leads to a hypothesis that in the wild songs needed to be simple to secure species identification, but under domestication this constrains was set free. [Work supported by JSPS Kakenhi #23118003.]

10:10–10:25 Break

10:25

4aAB5. What does Da Vinci have to do with it? Laurie L. Bloomfield (Psych., Algoma Univ., 1520 Queen St. E., Sault Ste. Marie, ON P6A2G4, Canada, laurie.bloomfield@algomau.ca)

It may seem unusual to suggest a relationship between Leonardo Da Vinci (1452–1519) and the study of avian communication, as he is perhaps better known for his attempts at developing a theory of human flight based on the principles of avian flight. His approach, however, was that “from the study of structure comes the knowledge of function.” Here, I present how an understanding of the structure of the various vocalizations produced by chickadees may lend to an understanding of the function of these vocalizations. Chickadees are an excellent model system for this type of research given that they produce various calls that are comprised of individual units that may function in different ways. We have conducted several bioacoustics analyses in the search for similarities and differences among call structures, as well as attempted to delineate the bioacoustical markers that would provide meaningful information to listeners. Further, I will discuss what constitutes human “language” and how the calls of chickadees may satisfy the criteria for a non-human language. With this in mind, we use various field and laboratory techniques in an attempt to understand the structure of vocalizations which may in turn convey information regarding the function of the vocalizations.

10:50

4aAB6. Extreme vocal plasticity in adult budgerigars: Analytical challenges, social significance, and underlying neurogenetic mechanisms. Timothy F. Wright (Biology Dept., New Mexico State Univ., Foster Hall, MSC 3AF, Las Cruces, NM 88003, wright@nmsu.edu), Erina Hara (HHMI Janelia Farms, Ashburn, VA), Anna M. Young (Otterbein Univ., Colombus, OH), Marcelo Araya Salas (Biology, New Mexico State Univ., Las Cruces, NM), Christine R. Dahlin (Univ. of Pittsburgh Johnstown, Johnstown, PA), Osceola Whitney (Biology, New Mexico State Univ., Las Cruces, NM), Esteban Lucero (Anschutz Medical Ctr., Univ. of Colorado Denver, Denver, CO), and Grace Smith Vidaurre (Biology, New Mexico State Univ., Las Cruces, NM)

Vocal learning is a complex trait that is expressed differently across different taxa. While many of the best-studied species exhibit close-ended learning, in which vocal signals are learned from adult tutors during juvenile critical learning periods, other species have open-ended learning in which new signals are learned throughout life. Budgerigars (*Melopsittacus undulatus*) are an extreme example of an open-ended learner, in which both sexes have a repertoire of multiple contact calls types that continually change through adulthood to match the call types of social associates. We discuss the analytical challenges posed by the rapid plasticity of the budgerigar vocal repertoire and compare results from several different approaches to characterizing call variation. We then consider the social implications and benefits of contact call matching in fission–fusion groups. Finally, we examine the mechanisms underlying this plasticity, with a special focus on the role of the gene *FoxP2*. We find downregulation of *FoxP2* mRNA and protein in the primary parrot vocal learning center, MMSt, across a variety of social conditions in which birds also show vocal plasticity. The results support the hypothesis that *FoxP2* is a key gene regulating the neural plasticity that underlies the persistent vocal plasticity exhibited by budgerigars.

11:15

4aAB7. Context-dependent categorical perception in a songbird. Stephen Nowicki (Biology, Duke Univ., 214 Biological Sci. Bldg., Durham, NC 27708, snowicki@duke.edu)

The division of continuously variable acoustic signals into discrete perceptual categories is a fundamental feature of human speech. Other animals have been found to categorize speech sounds much the same as humans do, although little is known of the role of categorical perception by animals in their own natural communication systems. A hallmark of human categorical perception of speech is that linguistic context affects both how speech sounds are categorized into phonemes, and how different versions of phonemes are produced. I first review earlier findings showing that a species of songbird, the swamp sparrow, categorically perceives the notes that constitute its learned songs and that individual neurons in the bird’s brain show categorical responses that map onto its behavioral response. I then present more recent data, using both discrimination and labeling tests, that show how swamp sparrows perceive categorical boundaries differently depending on context. These results demonstrate that there is a more complex relationship between underlying categorical representations and surface forms in the perception of birdsong. To our knowledge, this work suggests for the first time that this higher-order characteristic of human phonology is also found in a nonhuman communication system.

Contributed Paper

11:40

4aAB8. Vocal conditioning with playback of two template sounds in budgerigars. Yoshimasa Seki (Psych., The Univ. of Maryland, 1-1 Machihata-machi, Toyohashi 4418018, Japan, yoshimasa.seki@gmail.com) and Robert J. Dooling (Psych., The Univ. of Maryland, College Park, MD)

We examined a capability of budgerigars to produce a similar vocal pattern to a sound stimulus presented immediately just before. For this purpose, we trained birds using an operant conditioning procedure. In the training, two types of the birds' own call were used as the auditory stimuli. Then, they were tested by probe stimuli (another vocal pattern of the subjects' own, other birds' vocalization). At test trials, the birds vocalized not any similar sounds in response to the probe stimuli but one of the vocal patterns which was produced at the training trials. Then, we attempted to train the birds to produce vocal patterns following playback sounds which were slightly changing as the trials went. 24-step intermediate sounds between two birds' own vocal patterns were synthesized. Those intermediate sounds were shifting step-by-step from one to the other at each trial in a single session. Eventually, a bird created some novel sounds which were similar to those intermediate stimuli and had been never produced at the training trials. Taken together, the birds did not use the playback sounds as the vocal reference under the operant procedure. However, birds might store those playback sounds as the potential vocal repertoire.

THURSDAY MORNING, 5 NOVEMBER 2015 CLEARWATER, 8:30 A.M. TO 11:55 A.M.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Numerical and Analytical Modeling of Medical Ultrasound I

Martin D. Verweij, Cochair
Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, Lorentzweg 1, Delft 2628CJ, Netherlands

Robert McGough, Cochair
Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824

Chair's Introduction—8:30

Invited Papers

8:35

4aBA1. Simulation of therapeutic ultrasound treatments using the hybrid angular spectrum technique. Douglas Christensen (Bio-Eng., Univ. of Utah, 50 So Central Campus Dr., Salt Lake City, UT 84112, christen@ee.utah.edu), Scott Almquist (School of Computing, Univ. of Utah, Salt Lake City, UT), Alexis Farrer (BioEng., Univ. of Utah, Salt Lake City, UT), Dennis Parker, and Allison Payne (Radiology, Univ. of Utah, Salt Lake City, UT)

Numerical simulations play an important role in therapeutic ultrasound treatments. Simulations can help with transducer design, retroactively analyze temperature patterns for insight into treatment effectiveness, and ultimately be used for patient treatment planning. Our group has developed a rapid 3D simulation tool for ultrasound beam propagation named the hybrid angular spectrum (HAS) method. HAS is an extension of the traditional angular spectrum approach that uses fast Fourier transforms to alternate between the spatial frequency domain and the space domain as the beam propagates through inhomogeneous tissue regions. In this presentation, we briefly cover the physical and algorithmic principles underlying the HAS technique, then give examples of its use in two promising high-intensity focused ultrasound (HIFU) applications. First, we employ it to retrospectively predict the heating efficiency of transcranial treatments of 14 patients undergoing treatment for essential tremor using a large phased-array transducer. Phase aberration of the beams caused by skull irregularities is a major effect and must be modeled carefully for accurate results. Second, we model the extent of the phase aberration to be expected in our group's recently developed MRI/HIFU breast treatment system and show a correlation with the degree of breast tissue inhomogeneity in the path of the beam.

8:55

4aBA2. Methods and applications for modeling of continuous-wave ultrasound fields. T. Douglas Mast (Dept. of Biomedical, Chemical, and Environ. Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Techniques are presented for efficient and accurate computation of three-dimensional, continuous-wave radiated fields from ultrasound transducers of canonical geometries. Fields of flat circular pistons are computed using exact series expressions based on orthogonal function expansions of the Rayleigh integral. Fields of concave circular pistons are computed using analogous series expansions for an integral of effective source contributions over the radiating surface; this approach accurately approximates radiated fields of many focused ultrasound transducers of interest. Fields of flat and focused rectangular transducers are computed using closed-form expressions based on a modified Fresnel approximation; accurate fields are obtained at distances large compared to the radiating aperture dimensions, so that the near field for large transducers can be accurately simulated by subdividing the radiating aperture. Efficient methods for field computation facilitate design optimization for therapy and imaging devices and methods. In simulations of therapeutic ultrasound, local heat deposition is proportional to the squared magnitude of the radiated acoustic pressure. For pulse-echo imaging, beamformed echo signals can be simulated by appropriate convolutions of transmit-receive beam products with a three-dimensional model of the scattering medium. Illustrative examples include simulation of ultrasound thermal ablation, passive cavitation imaging, and pulse-echo B-mode and echo decorrelation imaging.

9:15

4aBA3. Four interpretations of temporal memory operators in the wave equation. Sverre Holm (Dept. of Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316, Norway, sverre@ifi.uio.no)

Attenuation and dispersion in ultrasound and elastography may be modeled with convolution memory operators in the wave equation. When the attenuation follows an arbitrary frequency power law the memory is a time domain power law. Since that is also a fractional time derivative, a large body of literature on fractional derivatives then becomes available for analysis and simulation. It can also be shown that, e.g., an elementary grain shearing process in an unconsolidated saturated sediment results in fractional wave equations for both compressional and shear waves. Second, some of these wave equations can be derived from constitutive equations with memory operators, ensuring satisfaction of causality and the Kramers-Kronig relations. The fractional Kelvin-Voigt and Zener models and their resulting attenuation and dispersion will in particular be discussed. The fractional operators can also be interpreted as the result of an infinite number of basic processes. Therefore, third, a fractional wave equation can be shown to be the result of an infinite number of elementary relaxation processes. Fourth, the fractional constitutive equation can also be expressed as an infinite sum of integer order derivatives of higher order which is also equivalent to a wave equation with higher order derivative terms.

Contributed Paper

9:35

4aBA4. Modeling geometrically complex layered regions in ultrasound using a modified T Matrix approach. Gregory T. Clement (Cleveland Clinic, 9500 Euclid Ave./ND 20, Cleveland, OH 44195-0001, gclement@physics.org)

A penetrable finite region consisting of layers of arbitrary shape, density, and sound speed is considered. Solutions are formulated as a T-matrix operator that is independent of the incident field, thereby allowing scattering due to any external source field to be rapidly calculated in a single operation. In departure from standard methods, boundary terms are written in their Fourier integral forms in polar coordinates. It will be shown that the approach allows scattering terms to be reduced to a single integral equation, a set of which form a block tri-diagonal matrix; a well-studied matrix form with various methods available for efficiently solving the inverse problem. A numeric algorithm is developed around the approach and validated against a previously reported wave-vector approach. Several medically relevant examples will be shown and discussed along with the potential for such methods to aid in the development of diagnostic and therapeutic techniques. [Study funded under National Institutes of Health Grant R01EB014296.]

Invited Papers

9:50

4aBA5. Next generation full-wave ultrasound simulation: Exploiting parallelism in the move toward exascale. Bradley Treeby (Medical Phys. and Biomedical Eng., Univ. College London, Biomedical Ultrasound Group, Wolfson House, 2-10 Stephenson Way, London NW1 2HE, United Kingdom, b.treeby@ucl.ac.uk), Jiri Jaros (Faculty of Information Technol., Brno Univ. of Technol., Brno, Czech Republic), and Ben Cox (Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom)

In recent years, the capability of performing 3D full-wave nonlinear ultrasound simulations in tissue-realistic media has been demonstrated by a number of groups. These models have since underpinned a range of interesting studies into the interaction of ultrasound fields with the human body. However, simulations have thus far been limited to domain sizes on the order of 1 billion grid points. This limitation usually means either the spatial area or highest frequency of interest is truncated. Looking toward the exascale era, where supercomputers integrating over 1M cores are predicted to appear before 2020, there is a significant opportunity to use models to gain new insight into ultrasound-tissue interaction with unprecedented detail. The challenge is to develop suitable numerical methods that map to these massively parallel architectures, in particular, that minimize data movement, allow the exploitation of co-processors, and minimize the accumulation of numerical errors. Here, we show a novel domain decomposition approach for the Fourier collocation spectral method that allows ultrasound simulations to be distributed across up to 32k CPU cores or hundreds of accelerators with reasonable efficiency. Using this model, we demonstrate for the first time the possibility of ultrasound simulations exceeding 70 billion grid points.

10:10–10:25 Break

10:25

4aBA6. Ultrasound imaging of the coefficient of nonlinearity. Libertario Demi, Ruud J. van Sloun (Lab. of Biomedical Diagnostics, Eindhoven Univ. of Technol., Den Dolech 2, Eindhoven 5612 AZ, Netherlands, l.demi@tue.nl), Caifeng Shan (Philips Res., Eindhoven, Netherlands), Martin D. Verweij (Lab. of Acoutical Wavelfield Imaging, Delft Univ. of Technol., Delft, Netherlands), and Massimo Mischi (Lab. of Biomedical Diagnostics, Eindhoven Univ. of Technol., Eindhoven, Netherlands)

Cardiac ablation (CA) is increasingly used to treat atrial fibrillation. However, long-term success is relatively low, and the procedure carries serious risks. To this end, we are developing a β (coefficient of nonlinearity) imaging method that may be employed to perform both tissue characterization and real time temperature estimation to respectively plan, monitor, and execute optimal CA. Starting from a one-dimensional generalized form of the Westervelt equation, we derived an analytical procedure for extracting β which is then further adapted to echo-mode. To evaluate the method performances, *in-silico* and *in-vitro*experiments were performed. First, one- to three-dimensional simulations including linear array scanning of three-dimensional objects were obtained with the INCS method. Next, the ULA-OP scanner was used with an Esaote LA332 linear-array to image a phantom consisting of 2-layers obtained as a mixture of oil, gelatin and water. Varying the percentage of oil (β similar to fat), different β values were obtained for the two layers. Both *in-silico* and *in-vitro* results show the capability of the method to estimate β variations. Compared to existing methods, the proposed approach provides more stable estimations (spatially) and does not require a special transducer or set-up, being more easily applicable in a clinical setting.

10:45

4aBA7. Characterization of medical ultrasound fields using modeling with a boundary condition obtained from measurements. Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ., 1013 NE 40th St., Seattle, Washington 98105, va.khokhlova@gmail.com), Petr Yuldashev, Pavel Rosnitskiy, Maria Karzova, Oleg Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Adam Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan Cunitz, Michael Bailey, Lawrence Crum, and Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Numerical modeling is becoming an important metrological tool for accurate characterization of ultrasound fields generated by medical transducers in water and in situ. While acoustic propagation equations are well established, setting a boundary condition relevant to the experiment still remains a critical problem. Two methods differing in complexity are described to address this problem. The first method comprises 3D simulations of the Westervelt equation with a boundary condition determined from acoustic holography measurements. The second, simpler method utilizes simulations based either on the KZK or Westervelt equation with an equivalent source boundary condition obtained by matching linear simulations to low-amplitude beam profiles along and transverse to the axis of the source. Calculations with both methods are compared to fiber optic hydrophone measurements of 2D phased therapeutic arrays, a 128-element diagnostic probe, and strongly focused single-element transducers. It is shown that the 3D Westervelt model with holographic boundary condition can accurately simulate the entire nonlinear ultrasound field. The simplified methods based on an equivalent source are shown to give accurate results in the focal zone of the transducers, even when shocks are present in the focal waveform. [Work supported by NIH EB7643, EB016118, DK043881, and RSF 14-12-00974.]

11:05

4aBA8. Angular spectrum methods for nonlinear ultrasound wave propagation: Mathematical development and implementation. François Varray, Olivier Basset, and Christian Cachard (Creatis, Université de Lyon, 7 av jean capelle, Villeurbanne 69621, France, francois.varray@creatis.insa-lyon.fr)

The use of simulation tools is required in many domains, especially in medical ultrasound (US). In US propagation, the nonlinear wave distortion is used in clinical application as harmonic or contrast imaging. Various methods exist to compute and evaluate the harmonic increase depending on the medium and the probe geometry. Among these methods, the angular spectrum method (ASM) proposes good compromise between accuracy and computation time. After expressing the wave propagation equation into Fourier domain, either a quasi-linear approximation or a slowly variation envelope approximation (SVEA) can be used to evaluate the harmonic increase which provide two tools. SVEA overcomes the quasi-linear limitation and evaluates the nonlinear frequency interaction during the propagation. A GPU implementation and optimization of both techniques allow a fast computation of the full nonlinear propagation. For both techniques, the computed fields are close and validate our proposed approach. Thanks to the GPU implementation, the computation is strongly reduced compare to Matlab or C version. Of course, some limitations in ASM are present, such as the one way computation of the US field, which did not allow the computation of reflection for media with inhomogeneities in density, celerity, attenuation, or coefficient of nonlinearity.

Contributed Papers

11:25

4aBA9. Computed tomography-based aberration correction for transskull acoustic focusing: Comparison of simulations and measurements with focused ultrasound brain systems. Ryan M. Jones (Dept. of Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Yuexi Huang (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), Daniel Pajek, and Kullervo Hynynen (Dept. of Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Clinical focused ultrasound brain systems currently employ computed tomography (CT)-based phase and amplitude corrections to mitigate skull-induced distortions and restore an acoustic focus at the intended target. In this study, acoustic measurements were conducted using two transcranial magnetic resonance-guided focused ultrasound systems (ExAblate 4000, InSightec, Haifa, Israel) operating at 230 and 650 kHz. The acoustic fields generated by the devices within intact, water-filled *ex-vivo* human skulls near the geometric focus were mapped using a 0.5 mm diameter needle hydrophone. In addition, the signals transmitted from each individual array element were captured at various locations within the skull cavity. These measurements were repeated without the presence of the skull in order to determine the element-specific aberrations induced by the cranial bone for each target position. The experimental measurements were simulated using three previously developed transcranial ultrasound propagation models: an analytical method similar to that currently employed by clinical brain systems, a multi-layered ray-acoustic approach, and a three-dimensional full-wave propagation model based on the Westervelt equation. We will present a comparison of the models based on their computational complexity as well as their ability to both predict the phase and amplitude aberrations induced by the skull and reproduce the measured *in-situ* pressure field distributions.

11:40

4aBA10. Comparison between computational and experimental methods for the characterization of therapeutic ultrasound fields. Subha Maruvada, Yunbo Liu, Joshua E. Soneson, Bruce A. Herman, and Gerald R. Harris (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

Analytical modeling of medical ultrasound fields has been developed by the FDA to aid in pre-clinical characterization of therapeutic ultrasound devices. In order to assess this publicly available software, called the HIFU Simulator, acoustic and thermal measurements of power, pressure/intensity and temperature distribution have been performed for comparison. Measurement and modeling issues include using hydrophones and radiation force balances at therapeutic power levels, validation of simulation models, and tissue-mimicking material (TMM) development for temperature measurements. To better understand these issues, a comparison study was undertaken between simulations and measurements of the HITU acoustic field distribution in water and TMM, and temperature rise in TMM. For the specific conditions of this study, the following results were obtained. In water, the simulated values for p+ and p- were 3% lower and 10% higher, respectively, than those measured by hydrophone. In TMM, the simulated values for p+ and p- were 2% and 10% higher, respectively, than those measured by hydrophone. The simulated spatial-peak temporal-average intensity values in both water and TMM were greater than those obtained by hydrophone by 3%. Simulated and measured end-of-sonication temperatures agreed to within their respective uncertainties (coefficients of variation of approximately 20% and 10%, respectively).

THURSDAY MORNING, 5 NOVEMBER 2015 ORLANDO, 8:30 A.M. TO 11:40 A.M.

Session 4aEA

Engineering Acoustics: Acoustic Material Characterization Methods

Michael R. Haberman, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758

Invited Papers

8:30

4aEA1. Ultrasonic interrogation of tissues and tissue-mimicking materials with the aid of wideband hydrophone-based transducer characterization. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Ultrasonic measurements can provide useful information to characterize biologic tissues and tissue-mimicking materials. Quantitative accuracy may be improved by first characterizing spatial and frequency dependence of ultrasound transducer beams, which may be measured with hydrophones in a water tank. Accuracy of estimates of acoustic output detected by hydrophones may be enhanced by deconvolving hydrophone sensitivity from hydrophone voltage output. Full deconvolution requires knowledge of magnitude and phase of hydrophone sensitivity, which may be measured over frequency bands of 1–40 MHz using time delay spectrometry. Time-delay spectrometry may also be used to provide wideband through-transmission measurements of attenuation and sound speed in tissues and tissue-mimicking materials. Much information regarding tissue composition and structure may be obtained from (1) through-transmission mode measurements of attenuation, sound speed, and dispersion, (2) pulse-echo mode measurements of backscatter, and (3) pulse-echo tracking of shear waves induced by acoustic radiation force push beams.

8:50

4aEA2. Monitoring hardening of concrete using ultrasonic guided waves. Jinying Zhu and Hongbin Sun (Civil Eng., Univ. of Nebraska-Lincoln, 1110 S 67th St, PKI 204C, Omaha, NE 68182, jyzhu@unl.edu)

Evaluation of early age properties of concrete is critical for ensuring construction quality of concrete structures. Although ultrasonic wave based methods show potential for monitoring the hardening process of concrete in laboratory, there are many challenges related to sensor installation and data acquisition in practice. In this study, the authors present an ultrasonic guided wave method that uses guided waves in a steel rebar to monitor hardening of surrounding cement paste and mortar. The longitudinal L(0,1) mode guided wave in rebar is excited by an EMAT sensor and received by a piezoceramic P-wave ultrasonic transducer. Continuous measurements during cement hydration were used to monitor the longitudinal wave attenuation resulted from leakage from the rebar to the surrounding cement paste/mortar. Shear wave velocities in cement materials were also monitored at the same time. Experiments were performed on four cement paste samples and three mortar samples. Experimental results demonstrated a strong correlation between the guided wave leakage attenuation and shear wave velocity for all tested samples, and both parameters increase with the age of cement materials.

Contributed Papers

9:10

4aEA3. Characterization of visco-elastic material parameters by means of the ultrasonic polar scan method. Koen Van Den Abeele, Arvid Martens (Dept. of Phys., KU Leuven - Kulak, E. Sabbelaan 53, Kortrijk 8500, Belgium, koen.vandenabeele@kulak.be), Mathias Kersemans, Joris Degrieck (Dept. of Mater. Sci. and Eng., Ghent Univ., Gent, Belgium), Steven Delrue (Dept. of Phys., KU Leuven - Kulak, Kortrijk, Belgium), and Wim Van Paepegem (Dept. of Mater. Sci. and Eng., Ghent Univ., Gent, Belgium)

The Ultrasonic Polar Scan (UPS) is a non-destructive technique which insonifies a material spot on a sample using ultrasonic pulses from as many oblique incidence angles $\psi(\varphi,\theta)$ as possible. Mapping the transmitted time-of-flight (TOF) and/or amplitudes as function of the incidence angle $\psi(\varphi,\theta)$ in a polar representation yields a UPS image with intriguing patterns that represent a fingerprint of the local visco-elastic properties. The present paper reports on recent advances in the revival of the UPS technique, involving the construction of an automated high-precision UPS scanner, the implementation of advanced simulation models as well as the development of efficient inversion routines. On the experimental level, unprecedented high quality TOF and amplitude UPS landscapes for a range of orthotropic (fiber reinforced) materials have been obtained. Using numerical simulation models, it can be readily demonstrated that the TOF UPS landscapes are directly connected to the elastic properties of the material, while the signal magnitudes displayed in the amplitude UPS landscapes are merely determined by the viscosity. By developing a coupled inversion scheme using the two landscapes simultaneously, we can achieve a full determination of the visco-elastic tensor. The new advances and inversion scheme will be illustrated in the case of fiber-reinforced plastics.

9:25

4aEA4. Acoustical characterization of nano-porous carbons. Kirill Horoshenkov, Michael Pelegrinis (Mech. Eng., Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk), Marco Conte (Dept. of Chemistry, Univ. of Sheffield, Sheffield, United Kingdom), Rodolfo Venegas (Carbon Air Ltd, Salford, United Kingdom), and Olga Umnova (Univ. of Salford, Salford, United Kingdom)

The acoustical and related non-acoustical properties of activated carbon were studied to understand better the effect of added catalyst and acid treatment on the nano- and micro-structure of activated, porous nano-carbon. The acoustical impedance was measured in a 45 mm diameter impedance tube with a special adaptor to accommodate a very small material quantity available for this experiment. The presence of the adaptor was compensated using the procedure detailed in Dupont *et al.* [POMA 19, 065008 (2013); http://dx.doi.org/10.1121/1.4799701]. The acoustic impedance of activated carbon was predicted using the model proposed by Venegas [Section 6.1, Ph.D. thesis, University of Salford, 2011]. The micro- and nano-scale porosities and pore sizes were determined by fitting the model to the acoustic impedance data in the frequency range between 50 and 1000 Hz. It is shown that the acid treatment and addition of catalyst result in the reduced radius of nano-pores and reduced nano-porosity. These effects are small but measurable acoustically. These results are consistent with the results of the BET experiment. This work provides the foundation for the development of acoustical methods for nano-porous material characterization which are rapid and non-invasive.

9:40

4aEA5. Acoustic characterization of flexural metamaterial elements using an impedance tube. Matthew D. Guild, David Calvo, and Gregory Orris (NRC Res. Associateship Program, Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, matthew.guild.ctr@nrl.navy.mil)

Acoustic metamaterials have been a topic of interest in recent years and have enabled exotic effective fluid properties to be achieved, including those with negative or near-zero dynamic values. These extreme properties arise through the particular design of the microstructure, which make use of simple microscale acoustic elements to create the desired macroscopic characteristics. While many of these acoustic elements utilize the quasistatic motion of the surrounding fluid, recently thin elastic plates have been receiving more attention in the use of transmission-line acoustic metamaterials. Due to the high acoustic impedance and flexural coupling of the elastic material, the acoustic characterization of these elastic structures using traditional techniques such as an air-filled impedance tube presents a significant challenge. In this work, flexural elastic elements are examined in an acoustic impedance tube. Using theoretical formulations for elastic plates in conjunction with acoustic impedance tube measurements, information about both the effective and intrinsic material properties of the flexural metamaterial elements can be obtained. The results of this analysis and its implications on acoustic metamaterial design will be discussed. [Work supported by the Office of Naval Research and the National Research Council.]

9:55–10:15 Break

Invited Papers

10:15

4aEA6. Multimode nonlinear resonant ultrasound spectroscopy (NRUS): From one-dimensional to three-dimensional characterization of the hysteretic elastic nonlinearity. Timothy J. Ulrich, Marcel Remillieux, Pierre-Yves Le Bas (Geophys. Group, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, tju@lanl.gov), and Cedric Payan (Laboratoire de Mecanique et d'Acoustique, Aix-Marseille Univ., Marseille, France)

Nonlinear Resonant Ultrasound Spectroscopy (NRUS) has been used extensively over the last two decades to quantify, through the nonlinear parameter α, the hysteretic nonlinearity of materials for geophysical, biomedical, and civil engineering applications. This technique relies on the variations of the damping and frequency of a resonance mode with the amplitude of this mode. A typical NRUS experiment is conducted on a long bar using its first longitudinal mode. In some experiments, higher order modes have been used because the nonlinearity was more pronounced but the type of motion involved has not been characterized. The parameter α measured from these experiments is then used to calibrate a 1D model of the non-classical nonlinearity. As a first step toward extending this model from 1D to 3D, experiments were conducted on long bar samples (assumed to be macroscopically isotropic) where modes are excited selectively and the type of motion involved in each mode is well characterized. In this simple isotropic case, longitudinal and torsional motions are decoupled in order to find the α_{11} and α_{44} parameters that correspond to the compression (C_{11}) and shear (C_{44}) moduli, respectively.

10:35

4aEA7. Laboratory measurements of compressional and shear wave properties in reconstituted mud and comparison to marine sediment acoustic propagation models. Kevin M. Lee, Megan S. Ballard, Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Laboratory measurements were performed to characterize propagation of compressional and shear waves in mud, whose composition differs significantly from granular marine sediments like sand or silt. Muddy sediments are colloidal suspensions of thin, irregularly shaped platelets that carry surface charges linked to their cation exchange capacities. These suspensions result in flocculent structures, which cause mud to have high porosity and exhibit gel-like behavior. Samples of reconstituted mud were prepared by mixing kaolinite powder with distilled water and then placing them under vacuum to remove air bubbles entrained by the mixing process. Pairs of hydrophones and benders elements were immersed in the samples to measure compressional (50 kHz to 500 kHz) and shear (0.1 kHz to 1.5 kHz) wave speed and attenuation, respectively. A resonator tube technique was used to infer compressional wave speed at lower frequencies (1 kHz to 10 kHz). Wet-dry mass measurements characterized the density and porosity of different mud samples, and electron microscopy was used to estimate platelet size distributions. The measured material parameters were used as inputs to various sediment acoustic propagation models. The comparison between the predicted and measured wave speeds and attenuations will be described. [Work supported by ARL:UT and ONR.]

Contributed Papers

10:55

4aEA8. Modified resonator method for laboratory measurement of the low frequency compressional wave speed in granular sediments. Gabriel R. Venegas and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu)

Several models have been developed to describe dispersion in granular sediments. There is an abundance of measurements confirming these models above 10 kHz, obtained via time-of-flight measurements, but there is far less data for accurate model verification under 10 kHz. The present work focuses on laboratory compressional wave speed measurements of water-saturated glass beads in the frequency range 1 kHz to 10 kHz using a resonance tube technique that, in future work, will be scaled in size to attain results below 1 kHz. In previous versions of this technique, the granular sediment completely filled the resonator; however, grain interaction with the resonator walls and the development of grain-to-grain force chains yielded undesirable results. The present method isolates a cylindrical volume of sediment from the resonator wall by a tulle net membrane and a layer of water. The effective sediment wave speed is inferred through the measured system resonance frequencies and a finite-element model that relates the system resonance frequencies to the intrinsic sound speed of the sediment. Sound speed measurements in a model sediment composed of 1-mm-diameter glass beads and distilled water agreed with the Effective Density Fluid Model (EDFM). [Work supported by ONR.]

11:10

4aEA9. Numerical and experimental analysis for the construction of an *in situ* measurement system of the acoustic impedance to be used as an academic tool. Lucas C. Lobato and Eric B. Carneiro (Structures and Construction, Federal Univ. of Santa Maria, Av. Roraima n° 1000, Santa Maria, Rio Grande do Sul 97105-900, Brazil, lucascostalobato@gmail.com)

Due to the need for a non-destructive method to measure sound absorption of acoustic materials, *in situ* methods have achieved significance in research on acoustic. So, the goal of this work is to build and test a system of *in situ* measurement of acoustic impedance to be used as an academic tool for undergraduate students of Acoustical Engineering on the Federal University of Santa Maria, Brazil. At first, two measurement techniques are introduced in this article: PP system and PU probe. Then, numerical analyses through a FEM model for both techniques are presented. The choice of FEM to model the measurement system occurs as an alternative to the BEM model, widely used in the literature. Thus, through the numerical model, errors were observed due to the finite size of sample. So, a strategy based on techniques proposed by the literature was applied, obtaining better precision. All numerical analysis were referenced on the Johnson-Champoux-Allard model of porous material. At last, a prototype of the measurement system is presented, followed by experimental results, using two microphones (PP system). There results were compared to measurement performed with an Impedance Tube.

11:25

4aEA10. Preliminary evaluation of the sound absorption coefficient of a thin coconut coir fiber panel for automotive applications. Key F. Lima, Nilson Barbieri, Fernando J. Terashima, Victor P. Rosa (Mech. Eng., PUCPR, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com), and Renato Barbieri (Mech. Eng., UDESC, Joinville, SC, Brazil)

Absorbent materials are fibrous or porous and must have the property of being good acoustic dissipaters. The noise reduction process occurs due to transformation of a part of the sound energy into heat. This process occurs when a sound wave propagates through pores or irregular arrangement of fibers. Sound propagation causes multiple reflections and friction of the air present in the absorbent medium transforming sound energy into heat. The acoustic surface treatment with absorbent material are widely used to reduce the reverberation in enclosed spaces or to increase the sound transmission loss of acoustic panels. In addition, these materials can also be applied to acoustic filters with the purpose to increase their efficiencies. The sound absorption depends on the excitation frequency of the sound and it is more effective at high frequencies. Natural fibers such as coconut coir fiber have a high potential to be used as sound absorbing material. Natural fibers are agriculture waste, manufacturing this fiber is a natural product, therefore an economic and interesting option. This work compares the sound absorption coefficient of a thin coconut coir fiber panel in relation to a composite panel made of fiberglass and expanded polyurethane foam used in the automotive industry. The evaluation of sound absorption coefficient was carried out with the impedance tube technique.

THURSDAY MORNING, 5 NOVEMBER 2015 GRAND BALLROOM 2, 9:00 A.M. TO 11:50 A.M.

Session 4aMU

Musical Acoustics: Stick-Slip Processes in Musical Instruments

Thomas R. Moore, Chair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—9:00

Invited Papers

9:05

4aMU1. Playability of a bowed string physical model including finite-width thermal friction and hair dynamics. Esteban Maestre (McGill Univ., Roc Boronat 138, Barcelona 08018, Spain, esteban@ccrma.stanford.edu), Carlos Spa (Univ Federico Santa Maria, Santiago, Chile), Quim Llimona, Gary P. Scavone (McGill Univ., Montreal, QC, Canada), and Julius O. Smith (Stanford Univ., Stanford, CA)

We report on the playability of a bowed-string physical model that combines digital waveguide and finite-difference time-domain frameworks. We extend previous approaches by combining a finite-width bow-string interaction model with a dynamic friction model based on simulating heat diffusion along the width of the bow. Bow hair dynamics are incorporated in the bow-string interaction, which includes two transversal string polarizations. The bridge termination is realized using a digital reflectance matrix model obtained from fitting two-dimensional driving-point admittance measurements. We present preliminary results from a playability study in which we explore the establishment of Helmholtz motion in static and dynamic bowing conditions.

9:25

4aMU2. Exploring the bowed string dynamical behavior using a linearized model approach. Vincent Debut (Instituto de Etnomusicologia - Centro de Estudo de Musica e Dança, Faculdade de Ciencias Sociais e Humanas, Universidade Nova de Lisboa, Lisbon 1069-061, Portugal, vincentdebut@ctn.ist.utl.pt), Octávio Inácio (Musical Acoust. Lab., NIMAE, School of Music and Performing Arts of the Polytechnic Inst. of Porto, Porto, Portugal), and José Antunes (Centro de Ciências e Tecnologias Nucleares, Instituto Superior Técnico, Universidade de Lisboa, Lisbon, Portugal)

For several decades bowed-strings have captured the attention of many researchers aiming for a thorough understanding of this system. Different approaches have been adopted particularly in the time-domain numerical simulations of the self-excited nonlinear regimes. Recently, the authors have been exploring the advantages of using a linearized approach to this problem, with or without the body coupling influence. Despite the highly non-linear bow/string friction force, the problem can be linearized about the average sliding velocity, as usually done in break-squeal noise, and an eigenvalue analysis can offer interesting information. For example, this approach allowed exploring the modal dynamics of bowed-string/body coupled system, studying the prediction of modes instabilities and the possible emergence of a strongly coupled mode responsible for the wolf-note, among other features. Here, using the linearized modal dynamics of bowed-strings we look in detail to the behavior of the string modes as a function of the bowing parameters. We start from a modal formulation of the string acted by the nonlinear bowing forces and develop the corresponding linearized formulation, which enables computation of the complex eigenvalues and eigenvectors as a function of the bowing velocity and normal force as well as location of the bow on the string.

9:45

4aMU3. Bifurcations in cello bowing using bow force below or above the Helmholtz regime. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de) and Mobert Mores (Dept. of Media Technol., Univ. of Appl. Sci., Hamburg, Germany)

Two bifurcation regimes with bowing forces above and below the thresholds for regular Helmholtz or sawtooth motion are investigated on a cello. During operation, string acceleration is measured at the bridge while bow force and speed are defined and measured on a bowing machine. High bow force causes subharmonics where the pitch depends on bowing pressure and velocity. Here sudden pitch changes were recorded at playing parameter thresholds while with slightly different parameters quasi-random scratchy sounds occurred. At very low bow forces, bifurcations appeared in the higher harmonics with two- and four-fold subdivision of periodicity, while the fundamental pitch remained stable. The reason for these bifurcations are bistable periodicities of the time series. To account for this, the sounds were analyzed using a Finite-Difference cochlea model which results in Interspike Intervals (ISI) with precise temporal resolution of these periodicities.

10:05

4aMU4. Maximum bow force revisited for the cello—Instrumentation with a precision pendulum. Robert Mores (Design Media Information, Univ. of Appl. Sci. Hamburg, Finkenau 35, 104, Hamburg 22081, Germany, robert.mores@haw-hamburg.de)

Schelleng (1971), Askenfelt (1989), Schumacher (1993), and Schoonderwaldt *et al.* (2008) formulated—in slightly different ways—how the maximum bow force relates to bow velocity, bow-bridge distance, string impedance, and friction coefficients. Related measurements at the respective transitions between Helmholtz und bifurcation regimes cover a diverse scenario of bowing machines and stringed instruments. So far, the empirical data does not clearly support either of the theories in a general way. A bowing pendulum of virtually infinite radius has been constructed to allow precise measurement of relevant bowing parameters. Two cellos are measured across all strings for three different bow-bridge distances. The empirical data suggest that linear relations predict the maximum bow force sufficiently well and a more distinct general model can be drawn. Furthermore, the pendulum employs an adaptive bow driving mechanism instead of a motor or engine. Such adaptive bowing discloses that mentioned regimes are stable and transitions between them sometimes require a hysteresis on force and speed variations. This explains some of the uncertainties in earlier studies and in this study. To confirm the findings the friction coefficients are measured separately by means of the same pendulum construction.

10:25–10:40 Break

10:40

4aMU5. Temporal phase evaluation of transients in the Brazilian cuíca drum. Tatiana Statsenko and Wilfried Kausel (Inst. of Music Acoust. (Wiener Klangstil), Univ. of Music and Performing Arts, Anton-von-Webern-Platz, 1, Vienna 1030, Austria, statsenko@mdw.ac.at)

Transient deformations can be observed by means of time-resolved electronic speckle pattern interferometry (trESPI), which implements a high-speed camera in the experimental setup and allows measurements of non-reproducible transients in real time. The stick-slip phenomena in musical instruments leads to transient nonlinear effects in the surface deformation, which can be resolved and characterized using trESPI. In this work, transient effects during the stick-slip motion of the Brazilian cuíca drum are investigated. Measurements of the cuíca under harmonic excitation and in playing conditions using trESPI are presented leading to a quantitative analysis of the motion.

11:00

4aMU6. Measurements of friction instruments with high-speed camera and subpixel tracking. Rolf Bader, Florian Pfeifle, Niko Plath, and Christian Koehn (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

Instruments working with stick-slip interactions not using a string were popular in the West since the invention of the glass harmonica by Benjamin Franklin in 1751. The Terpodion investigated was built by Buschmann in the mid-19th century as a friction instrument with a keyboard, where bars are pressed against a wooden rotating cylinder producing a sound. Using high-speed camera recordings, the only playable instrument today at the Viadrina museum in Frankfurt a.O. shows sinusoidal vibrations of the bars determining the played pitch while the radiated sound is highly complex. Therefore the instrument shows a fundamentally different stick-slip action compared to bowed instruments. Friction instruments of the East, like the singing bowls also show very sinusoidal-like sounds again caused by the stick-slip interaction. The New Ireland Lounuet, a finger-rubbed wooden block has both, a sinusoidal motion as well as a highly complex one which is to imitate bird and frog sounds. A systematic view on the different stick-slip interactions is suggested.

Contributed Papers

11:20

4aMU7. The Etiology of chatter in the Himalayan singing bowl. Chloe Keefer, Samantha Collin, and Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, ckeefer@rollins.edu)

The Himalayan singing bowl is a nearly symmetric idiophone played by rotating a wooden stick called a puja around the outer rim of the bowl. The vibrations of the bowl are excited by a stick-slip mechanism, which produces a radial motion of the bowl with a deflection shape similar to the (2,0) mode. We present experimental evidence that the position of the puja coincides with the point of minimum displacement on the bowl, indicating that it imposes a node in the deflection shape that rotates around the bowl with the puja. However, in many cases the puja is forced off of the bowl and an audible chatter is produced as the puja repeatedly strikes the bowl several times per second. This indicates that the position of the puja is not a node, but rather merely a point of minimum deflection. Examination of high-speed electronic speckle pattern interferograms and time-resolve acoustic spectra provide insight into the mechanics of the singing bowl and the origin of the chatter.

11:35

4aMU8. An unconditionally stable scheme for simulation of stick-slip processes. Vasileios Chatziioannou and Wilfried Kausel (Inst. of Music Acoust., Univ. of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, kausel@mdw.ac.at)

Stick-slip processes are often encountered in musical instruments, the most notable example being that of a bowed violin string. Modeling such nonlinear interactions has been often attempted in the field of Musical Acoustics, using a variety of time-stepping algorithms. The nonlinear nature of such problems requires careful analysis of the stability properties of the developed numerical schemes. This is usually carried out using energy based methods, in which case one needs to consider the continuous exchange of energy between the bow and the bowed object. In this paper an unconditionally stable scheme is proposed for the simulation of a bowed, lumped mass. The stability of the algorithm is ensured due to the presence of an invariant quantity. The numerical results are in agreement with well established algorithms, with the proposed methodology possessing no stability constraints and being extensible to a wide range of (lumped and distributed) acoustic systems.

THURSDAY MORNING, 5 NOVEMBER 2015 GRAND BALLROOM 1, 8:30 A.M. TO 11:40 A.M.

Session 4aNS

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: Thoughts on the Next Generation of ANSI Loudness Standards

Patricia Davies, Chair

Ray W. Herrick Labs., School of Mechanical Engineering, Purdue University, 177 South Russell Street, West Lafayette, IN 47907-2099

Invited Papers

8:30

4aNS1. How should we move forward with the next version of the ANSI S3.4 loudness standard? Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

Recent updates to the international loudness standard ISO 532 reflect both increased understanding of loudness and the use of loudness standards by the acoustics and engineering communities. Both are inspiration to consider revising ANSI S3.4-2007. The current ANSI standard is based on Moore and Glasberg's model of loudness for stationary sounds. The new ISO 532-2 is a revision of this. The new ISO 532-1 is based on Zwicker's time-varying loudness, and will replace the current ISO 532B, also based on Zwicker's loudness model but for stationary sounds. For people who are not developing loudness models but are using them to assess sounds or noise, there are a number of other issues, e.g., loudness of very low frequency sounds, and exposure assessment based on loudness. These will be discussed and possible directions in updating the current ANSI standard are presented.

8:50

4aNS2. Differences between ANSI S3.4-2007 and the proposed ISO532-2. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The ANSI standard for the calculation of loudness (ANSI S3.4-2007) is based on the model developed by Moore, Glasberg and co-workers, and uses the assumption that a diotic sound is twice as loud as that same sound presented monaurally. However, recent data suggest that loudness summation across ears is less than assumed in the ANSI standard. This was taken into account in the revised loudness model of Moore and Glasberg [J. Acoust. Soc. Am., **121**, 1604–1612, 2007] using the concept of "binaural inhibition", whereby the signal at one ear inhibits the internal response to a signal at the other ear. This model predicts that a diotic sound is 1.5 times as loud as that same sound presented monaurally, and it forms the basis for the proposed ISO532-2 standard. The revised model also gives reasonably accurate predictions of loudness in cases where the sounds differ across the two ears, for example, sounds recorded via a dummy head. An extension of the model to deal with time-varying sounds has been shown to give accurate predictions of loudness for a variety of technical and musical sounds. It is proposed that the revised and extended model be used as the basis for a new ANSI standard.

9:10

4aNS3. Determining binaural summation for stationary signals across the frequency spectrum. Colin J. Novak and Jeremy Charbonneau (Mech., Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Dept. of Mech. Eng., Windsor, ON N9B 3P4, Canada, novak1@uwindsor.ca)

The concept of binaural summation has been widely accepted; however, the value of summation, and the manner in which it is applied, is still under a great deal of debate. This research is an investigation of the binaural summation mechanisms through the use of a loudness comparison experiment for pure tones. Preliminary results have concluded that for pure tones presented at 40 dB the amount

of summation increases with frequency from approximately 2 dB at 40 Hz up to 8 dB at 10 kHz. Due to the low hearing threshold at 40 dB, the experiment was modified such that the hearing threshold of each individual is collected in order to presented each signal at a fixed amplitude above the threshold of hearing for each frequency (i.e., a sensation level of 20 dB(SL)). However, similar outcomes have been found using this approach, thus supporting the conclusions that binaural summation does exist and is found to increase at defined increments with increasing frequency for pure tones. The eventual outcome of this work is to improve the existing loudness metrics to allow for the input of binaural measurements taken using binaural mannequins.

9:30

4aNS4. Status quo of standardizing loudness of time-varying sounds. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Recently, a new ISO standard for loudness of arbitrary sounds ISO 532-1 (Zwicker method) was proposed for the revision of ISO 532:1975 (method B). The new standard is based on DIN 45631/A1:2010, which includes the widely used standard DIN 45631:1991 for stationary sounds as a special case. DIN 45631:1991 differs slightly from ISO 532:1975 (method B) by specifying corrections for low frequencies and by restricting the description of the approach to numerical instructions only, thus allowing a unique software description. ISO 532-1 eliminates uncertainties of existing standards by strictly defining the complete procedure of loudness calculation starting with the waveform of the time signal and ending with specific and total loudness vs. time functions. The strict definition of the complete procedure, given not only by formulae and tables but also by program code, is a step forward to comparability of calculated loudness results. ISO 532-1 shall update the previous ISO 532:1975 (method B) and adapt it to proven new practice while preserving procedural and database continuity. The method according to Moore/Glasberg based on the American standard ANSI S3.4-2007, for stationary sounds only, shall replace ISO 532:1975 (method A) and will be named as ISO 532-2 in the updated version.

9:50

4aNS5. Loudness of temporally varying environmental sounds. Jesko L. Verhey, Jan Hots (Dept. of Experimental Audiol., Otto von Guericke Univ. Madgeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de), Moritz Wächtler, and Jan Rennies (Cluster of Excellence Hearing4all Project Group Hearing, Speech and Audio Technol., Fraunhofer Inst. for Digital Media Technol. IDMT, Oldenburg, Germany)

Loudness of speech, speech-like signals, and other dynamic environmental sounds were measured and compared to predictions of current loudness models and predictions on the basis of the relevant standards. Loudness was assessed experimentally by using a loudness matching procedure and categorical loudness scaling. The scaling method used in our study is in agreement with the requirements of the international standard on categorical loudness scaling. Categorical scaling allows for a fast assessment of loudness over a large level range but has the disadvantage that loudness is not measured in sones, as commonly used in loudness models. The data of the two methods are compared by deriving levels at equal loudness from the categorical loudness data. In addition, the present study discusses to what extend the scaling data can be compared to loudness predictions by using recently proposed equations relating categorical units to sones. The comparison of the measured levels at equal loudness and simulations revealed that for speech and speech-like signals the long-term spectrum largely determines the loudness of the sound. Dynamic models with short time constants tend to overestimate loudness. In general, this is also true for the other environmental sounds although for some technical signals discrepancies remain.

10:10–10:25 Break

10:25

4aNS6. Some factors affecting loudness measurement and prediction. Robert S. Schlauch, Edward Carney, Tzu-Ling J. Yu, and Heekyung J. Han (Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 115 Shevlin Hall, Minneapolis, MN 55455, schla001@umn.edu)

The current ANSI standard (ANSI S3.4, 2007) estimates the loudness of sustained sound, but many naturally occurring sounds have time-varying levels. Glasberg and Moore (2002) published a model for the prediction of time-varying sounds and this aspect should be considered as part of a revision to the ANSI standard. To assess the predictions of the dynamic model, loudness magnitude estimation functions were obtained for 24 listeners using pure tones (0.5 and 1.0 kHz), vowels, spondees, and speech-shaped noise (SSN) presented at levels from 40 to 90 dB SPL. Inferred equal-loudness levels from the fitted loudness functions were compared to the model predictions. The loudness model was qualitatively consistent with the behavioral data. The model predicted SSN to be louder than vowels and spondees which would be louder than tones; however, the model over-predicted the loudness differences. Possible explanations for the over-prediction include biases in loudness judgments, cognitive factors regarding learned expectations for loudness, assumptions regarding the free-field-to-headphone transfer function, and the model's over-prediction of spectral, loudness summation, a result found recently (Schlittenlacher *et al*., 2014) for the ANSI standard and DIN 45631.

10:45

4aNS7. On time-varying loudness and nonlinearly propagated sound. Andrew Marshall, Karl Oelschlaeger, and Emily Belzer (Southwest Res. Inst., 6220 Culebra Rd., San Antonio, TX 78238-5166, andrew.marshall@swri.org)

Time varying loudness models have been found to be highly correlated with subject judgments for many sources of noise. Among these sources are impulsive sounds such as sonic booms, blast noise, and thunder. Aside from the short temporal duration, all of these sound sources are distinctive in that each is of high enough amplitude to have nonlinear propagation effects occur, which change the temporal and spectral character of a signal over distance. Because of this, it is possible that differences in how a model handles spectral content may result in differences in loudness predictions over distance. A series of simulated nonlinear sources were propagated and its time-varying loudness calculated from both the Moore & Glasberg and Zwicker model. These predictions will be compared to examine how these models differ when the spectral content of the signal changes. Other parameters, including historical level metrics and those derived from loudness time histories, such as the loudness time derivative, will also be discussed.

Contributed Paper

11:05

4aNS8. Loudness and noise rating: How may standardization bridge the gap between science and engineering? Florian Völk (Bio-Inspired Information Processing, Technische Universität München, Boltzmannstraße 11, Garching 85748, Germany, florian.voelk@mytum.de)

Current noise regulations and standards, in Europe and to some extent also elsewhere, are primarily based on A-weighted sound levels. Most perceptual aspects are—if at all—taken into account by additional correction levels, for example, punishing tonality or impulsive content. On the contrary, instrumental loudness-prediction methods have been available for more than half a century and were first standardized 40 years ago. Against this background, the question arises why loudness—regardless of being calculated according to Zwicker or Moore and Glasberg—has not yet been accepted more widely in the field of noise control and related standardization. This contribution attempts to raise some possibly influential aspects, and to discuss them in the light of next-generation loudness standards. Advantages of stationary and time-varying loudness predictions over corrected weighted levels will be discussed and contrasted with potentially contradicting arguments. The discussion is primarily intended to provide arguments for the selection of future loudness-prediction standards. However, also a broader perspective is taken, looking at interrelations with other standards, algorithmically and regarding limiting values.

11:20–11:40 Panel Discussion

THURSDAY MORNING, 5 NOVEMBER 2015 — ST. JOHNS, 8:30 A.M. TO 11:40 A.M.

Session 4aPA

Physical Acoustics and Noise: Launch Vehicle Acoustics I: Acoustics of Launch Vehicles and Supersonic Jets

Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

Chair's Introduction—8:30

Invited Papers

8:35

4aPA1. Near-field acoustical array measurements of an impinging supersonic jet. Kent L. Gee, Tracianne B. Neilsen, Darren K. Torrie (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), Masahito Akamine (Graduate School of Frontier Sci., Univ. of Tokyo, Chiba, Japan), Koji Okamoto (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwa, Japan), Susumu Teramoto, Takeo Okunuki (Graduate School of Eng., Univ. of Tokyo, Bunkyo-ku, Tokyo, Japan), and Seiji Tsutsumi (Japanese Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

Impingement significantly alters the rocket plume in the near-launch pad environment, which in turn affects the acoustic radiation. Prior laboratory measurements of an unheated, Mach 1.8 ideally expanded jet impinging on 45-degree inclined flat plate were carried out using a microphone that was moved within a relatively dense grid [Akamine *et al.*, AIAA J. **53,** 2061–2067 (2015)]. A multi-institution collaboration by the authors was begun in order to conduct array measurements of the acoustic radiation from both impinging and free jets. Measurements comprised a total of 42 measurement channels located within 40 nozzle diameters. An array of two-dimensional microphone probes was placed so as to examine the transition from the hydrodynamic near field to the acoustic radiation. A scanning linear array of microphones and a stationary polar array were also designed to enable beamforming, cross correlation, and partial field decomposition. This paper describes the jet facility, experiment design, and initial analyses conducted on the data collected. [Measurements supported by the Japanese Society for the Promotion of Science.]

8:55

4aPA2. Effects of time-varying grain shape on combustion instability of a solid rocket motor. Taeyoung Park, Hunki Lee, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, pty0948@yonsei.ac.kr), and Dohyung Lee (Agency for Defense Development, Seoul, South Korea)

A common approach to modeling combustion instability in a rocket propulsion system is to express the pressure oscillation in a combustor as a superposition of acoustic modes with time-varying amplitudes. Here, the tacit assumption is that the acoustic modes themselves remain more or less the same over the entire burning time. However, in the case of a solid rocket motor, the grain shape changes significantly as combustion progresses. This can gradually alter the shapes and frequencies of the acoustic modes, the influence of which on combustion instability has rarely been discussed in the existing literature. In this study, the effects of time-varying grain shape are modeled by introducing a slow time scale associated with the progressive burning of the grain. The resulting model equation accounts for the evolution of acoustic modes as well as their growth/decay in amplitude. Predictions with and without the use of the slow scale are compared with respect to measurements of a static firing test.

9:15

4aPA3. Far-field acoustical measurements during a Space Launch System solid rocket motor static firing. Blaine M. Harker, Brent O. Reichman, Trevor A. Stout, Eric B. Whiting, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net)

Acoustical measurements were made in the very far field during a recent test firing of the five-segment QM-1 Space Launch System solid rocket motor at Orbital ATK. Data were taken using 6.35 mm and 12.7 mm type-1 microphones at three far-field locations to the sideline and aft of the nozzle at a range of 650–800 nozzle diameters. The experiment setup, including the appreciable terrain changes, is first discussed. Spectral and autocorrelation analyses highlight the variation of the noise with respect to observation angle. In addition, high-frequency spectral characteristics and waveform statistics are evidence of the significant nonlinear propagation over the propagation range. Effects of microphone size, terrain effects, and data stationarity during the firing are discussed. This dataset is compared to measurements of other solid rocket motors at closer and farther ranges, including the GEM-60 and the four-segment Shuttle Reusable Solid Rocket Motor.

9:35

4aPA4. Multiresolution non-stationary techniques for the characterization of transient pressure signals embedded in broadband plume noise. David Alvord and Alessio Medda (Aerosp. & Acoust. Technologies Div., Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.alvord@gtri.gatech.edu)

The acoustic environment generated by the Space Shuttle propulsion system historically was considered one of the most complex rocket noise environments to predict of NASA's heritage rockets. After the Space Shuttle was retired in 2011, the Space Launch System (SLS) architecture chosen to succeed the shuttle will generate an even more complex acoustic environment than Space Shuttle. Both Space Shuttle and SLS have dual first stage propulsion systems consisting of two Solid Rocket Boosters (SRBs) and three (Space Shuttle) or four (SLS) RS-25 LH2/LOX engines. The additional complexity for the SLS acoustic environment arises due to the in-line propulsion configuration (as opposed to the offset configuration for Space Shuttle) firing both SRB and liquid engine plume into the same exhaust duct at liftoff. For Space Shuttle, the offset configuration meant each acoustic phenomena was separable and could be predicted and analyzed individually. For SLS, the in-line configuration means the IOP waves (one per SRB) are injected directly into the steady state plume meaning the events can no longer be assumed separable. This paper discusses data from a scaled down configuration and applies wavelet analysis in attempt to extract and characterize an injected transient blast wave from steady state plume noise.

9:55–10:15 Break

10:15

4aPA5. Spatiotemporal correlation of high-performance military aircraft jet noise. Blaine M. Harker, Tracianne B. Neilsen, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Correlation analyses of pressure measurements on a ground-based array of microphones of noise from a tethered high-performance military aircraft provide insights into the sound field variation with position and engine conditions which are fundamental in the continued development of more complete jet noise models. Time-scaled, single-point (auto)correlation functions confirm that to the side of the nozzle exit, the temporal correlation envelope decays very rapidly, whereas the envelope decays more slowly in the maximum radiation region and farther downstream. Two-point space-time (cross) correlation functions confirm that noise from a single engine operating at intermediate power is more similar to that from heated, laboratory-scale jets, whereas additional features seen at military power and afterburner are unique, including a feature which is likely related to the dual directivity lobe observed in the far field of military aircraft. A complementary coherence analysis provides estimates of spatial coherence lengths as a function of frequency and location. Field coherence lengths are utilized in analyzing coherence lengths of equivalent source distributions obtained from applying DAMAS-C to the ground-based array data. The cumulative results of these investigations provide a full-scale military jet noise benchmark that should be considered when evaluating laboratory-scale jet studies and simulations of jet noise.

10:35

4aPA6. Inclusion of a ground-reflecting plane in wavepacket modeling of military jet noise. Tracianne B. Neilsen, Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Michael M. James (Blue Ridge Res. & Consulting, Asheville, NC), and Blaine M. Harker (Brigham Young Univ., Provo, UT)

Equivalent source models for high-speed jet noise are intended to represent the acoustic properties of the turbulent mixing noise. A wavepacket ansatz previously applied to obtain equivalent sources from anechoic laboratory measurements is modified to include both a direct and image complex pressure representation of the source to model the presence of a ground-reflecting plane. This is important for modeling the sound field at maintainer and flight deck personnel positions because the interference effects caused by ground-reflected propagation paths significantly influence the sound levels and vary with frequency and location. The ability of the direct-plus-image wavepacket model to yield the interference patterns observed across large planes of data (2 m tall by 23 m long) measured near a high-performance military aircraft is evaluated. In particular, the variation in the nulls of the sound pressure level across the planes as a function of frequency predicted by this wavepacket model are compared to the measurements between 4 and 25 m from the engine nozzle exit. Comparisons with previous work based on a Rayleigh-distributed source amplitudes are also provided. [Work supported by the Office of Naval Research.]

Contributed Papers

10:55

4aPA7. Quantitative nonlinearity in subsonic and supersonic model-scale jet noise. Kyle G. Miller (Dept. of Phys. and Astronomy, Brigham Young Univ., 323 East 1910 South, Orem, UT 84058, kglenmiller@gmail.com), Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Anthony A. Atchley (Gradate Program in Acoust., Penn State Univ., Provo, UT)

Understanding the impact of jet noise, including annoyance due to crackle, can be improved by quantifying the nonlinearity in a signal with a single-microphone measurement. An ensemble-averaged, frequency-domain version of the generalized Burgers equation has been used to find a quantitative expression for the change in sound pressure level spectrum, L_p, with distance, r, due to the separate effects of geometric spreading, absorption, and nonlinearity. The nonlinear term, based on the dimensionless nonlinearity indicator known as "Q/S," has been used to characterize the frequency-dependent nonlinearity as a function of angle and distance in subsonic (Mach-0.85), overexpanded (Mach-1.8), and ideally expanded (Mach-2.0) model-scale jet data. Analyses show that nonlinear effects in the Mach-2.0 data are about twice as strong as those in the Mach-1.8 data, but such effects are completely absent in the Mach-0.85 data. [Work supported by the AFRL SBIR program.]

11:10

4aPA8. Comparison of measured and predicted statistical measures in military jet noise propagation. Brent O. Reichman (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Alan T. Wall (Air Force Res. Labs, Wright-Patterson Air Force Base, OH), Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Crackle, an annoying component of jet noise, has been associated with acoustic shocks that form as a waveform experiences nonlinear steepening. The skewness of the first time derivative of the pressure waveform, or derivative skewness, is a metric that is sensitive to acoustic shocks, and has shown that shocks, and therefore crackle, are found in the far field of full-scale military jets. Recent extensive measurements on F-35 aircraft enable a comparison of linear and nonlinear predictions and measured waveforms over a large spatial area. Waveforms measured at 76.2 m from the aircraft are numerically propagated using linear and nonlinear methods to several distances up to 305 m, and the derivative skewness calculations from these numerically propagated waveforms are compared with those of measured signals. Comparisons are made for both the A and B variants of the F-35, in changing meteorological conditions, and with engine conditions ranging from intermediate power to full afterburner. [Work supported by USAFRL through ORISE.]

11:25

4aPA9. Estimating aircraft noise levels and spectra from aircraft flyover and ground operations. Alan T. Wall and Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com)

The noise from military aircraft operations can adversely affect people within the civilian community near an airbase and within the military community living and operating within the confines of the airbase. The aircraft noise levels and spectra should be estimated to determine the building noise attenuation treatments to achieve the desired interior noise levels. This presentation describes a method to estimate the noise levels and spectra from both ground operations and flyover operations of high performance military jet aircraft for the community near the airbase and the airbase proper. Sound levels are predicted over a typical airbase region using an equivalent source model for a fighter jet aircraft and atmospheric propagation modeling. Overall level contours are generated over the entire area, and spectral levels are predicted at desired points corresponding to noise sensitive building locations. [Work supported by USAFRL through ORISE.]

Session 4aSC

Speech Communication: Development of Speech Production and Perception Across the Lifespan

Mary E. Beckman, Cochair
Linguistics, Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210-1298

Valerie Hazan, Cochair
Speech, Hearing and Phonetic Sciences, UCL, Chandler House, 2, Wakefield Street, London WC1N 1PF, United Kingdom

Invited Papers

8:50

4aSC1. Clear speech adaptations across the lifespan. Valerie Hazan (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

Talkers need to be able to alter their speech production in order to communicate effectively in conditions in which there are acoustic or linguistic barriers to communication. The clear speaking styles that result from such adaptations have been well documented, but mostly for young adult talkers. As a skilled aspect of speech production, the ability to make such adaptations may develop late during acquisition and be affected by reduced motor or cognitive control in older talkers. In a series of studies, we are investigating how speech adaptations in challenging communicative conditions change across the lifespan: in children aged 9 to 15 years, in young adults, in older adults aged 65 and above. In all three studies, recordings were made while talker pairs complete a cooperative problem-solving task (diapix) when communication was easy or when a communication barrier was placed on one talker. Acoustic analyses have shown adaptations to speaking rate, intensity, fundamental frequency characteristics, vowel formant space. While young adults seem adept at making adaptations that are well suited to overcome the specific interference experienced by their interlocutors, younger talkers appear to make less nuanced adaptations. We will discuss factors affecting such speech adaptations across the lifespan.

9:10

4aSC2. Speech perception and spoken word recognition in young children. Jan Edwards and Tristan Mahr (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, jedwards2@wisc.edu)

Recent studies have found that expressive vocabulary size predicts spoken word recognition in young children. Children with larger vocabularies recognize even highly familiar words more efficiently than their peers with smaller vocabularies (Fernald *et al*., 2006). Furthermore, lexical processing efficiency at 18 months predicts future vocabulary size up to 8 years of age (Fernald *et al*., 2013; Marchman & Fernald, 2008). However, these studies did not include any measures of speech/language development beyond vocabulary size. We discuss the results of several studies that included multiple measures of speech and language development as well as measures of the home linguistic environment (using LENA, Ford *et al*., 2008). One experimental task assessed children's online responses to correct productions of familiar words, to mispronunciations of these words, and to nonwords using eye-tracking. We found that speech perception was a better predictor of lexical processing efficiency than expressive vocabulary size. Furthermore, the number of conversational turns between children and their caregivers also predicted how quickly children looked at unfamiliar objects when they heard nonwords. Implications of these results are discussed. [Research supported by NIDCD grant 02932.]

9:30

4aSC3. Speech perception and production in sequential bilingual children: A longitudinal study. Kathleen McCarthy (Speech, Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, kathleen.mccarthy@ucl.ac.uk)

The majority of bilingual speech development research has focused on children who acquire both languages simultaneously from birth. Yet, for many bilingual children growing up in dense immigrant communities, their language experience is typically very different. These children, often referred to as sequential bilinguals, are initially exposed to their heritage language (L1) and in many cases are only immersed in the host country's language (L2) when they start school at around 4-years-old. To date, little is known about the developmental trajectory of sequential bilinguals. The current study tracked the acquisition of the English voicing contrast and monophthongal vowels by Sylheti-English sequential bilingual children from the London-Bengali community. Children were tested during preschool (mean age: 54 months old) and again one year later. The sequential bilinguals perception and production was initially driven by their L1 experience, resulting in less refined English phonemic categories than their monolingual peers. With L2 experience, children acquired the target phonemic contrasts, however they displayed a preference Sylheti phonotactics in a non-word repetition task. These findings have implications for our understanding of language development in complex multilingual settings, and will be discussed in light of the children's language input and broader phonological development.

9:50

4aSC4. Cognitive and linguistic influences on speech motor development. Ignatius Nip (School of Speech, Lang., & Hearing Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-1518, inip@mail.sdsu.edu)

Speech development is affected by the interaction of the development of various domains, including motor control and language. Infants demonstrate strong associations between speed and range of movement of the lips and jaw with cognitive (attention, memory) and language (number of words understood/expressed, number of gestures used) skills during language acquisition (Nip *et al*., 2011). In addition, goal-directed vocal behaviors (e.g., words) are produced with faster lip and jaw speeds in infants (Nip *et al*., 2009). Similar task-related changes can be observed in older children. Tasks requiring greater language formulation demands (e.g., re-telling stories) are produced with greater movement speeds and oral excursions than tasks requiring fewer demands (e.g., repeating syllables) (Nip & Green, 2013). Kinematic descriptions of speech production may provide insight into the acoustic consequences of adapting movement strategies for different tasks; increasing oral excursions may be a strategy to increase articulatory precision and may account for the reduced speaking rate during speaking tasks requiring greater demands (Nip & Green, 2013). In addition, coordination of articulatory movements is highly associated with intelligibility in children with and without speech disorders (Nip, in press). Understanding the interactions among motor control and language may explain patterns of speech development.

10:10–10:25 Break

10:25

4aSC5. The influence of gender identity on children's production of sibilant fricatives. Benjamin Munson (Univ. of Minnesota, 115 Shevlin hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, benjamin.ray.munson.jr@gmail.com)

Phonetic differences between men and women's speech are the consequence of sex-related variation in the anatomy of the speech-production system, as well as learned behaviors specific to particular languages and to specific social and cultural contexts. Children learn socially and culturally specific gendered phonetic variants concurrent with developmental changes in the vocal tract which give the child an increasing capacity to produce systematic phonetic variation. The research in this presentation is part of a larger project examining phonetic variation in 5–13 year old boys with gender dysphoria (GD). This presentation focuses on the acoustic characteristics of /s/, which was chosen because it is the locus of a great deal of gender-related phonetic variation that appears not to be the consequence of sex differences in vocal-tract size and shape. Preliminary analyses of a subset ($n=30$) of talkers from a corpus of 5 to 13 year old children ($n=104$, including boys with GD and boys and girls without GD), found that boys with GD produce more [θ]-like /s/ tokens than boys without GD in single words. This talk presents the results from the entire corpus, which allows for a more robust analysis of the influence of gender identity on children's fricative production.

10:45

4aSC6. Phonetic and lexical influences on changes across the lifespan. Jonathan Harrington and Johann U. Reubold (Inst. of Phonet. and Speech Processing (IPS), Univ. of Munich, Schellingstrasse 3, Munich 80799, Germany, jmh@phonetik.uni-muenchen.de)

The study considers longitudinal studies over several decades within the same individual in order to determine whether phonetic sound change takes place initially in more frequent words. The focus of the analysis was on the vowels of the Christmas Broadcasts by Queen Elisabeth II over several decades and in Alistair Cooke's Letter from America broadcasts. For the first of these, a re-analysis of the phonetic lowering of the vowel in the lexical set TRAP and of tensing of final lax vowel in HAPPY showed no effect of lexical frequency on sound change. The focus of analysis in the second was during a period in which the speaker was shown to acquire General American characteristics after emigrating to the United States from Britain, but then in later life to revert over a 5–10 year period back toward characteristics of British English Received Pronunciation. This reversion is shown to take place at a faster rate in lexically frequent words. While the evidence overall for lexically gradual changes is equivocal, the changes in both speakers are most appropriately modeled as a leveling due to variation in dialect contact: increasingly with middle class speakers for the Queen; decreasingly with American English speakers for Cooke.

11:05

4aSC7. Effects of age-related hearing loss on verbal processing and short-term memory. Esther Janse (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, PO Box 310, Nijmegen 6500 AH, Netherlands, e.janse@let.ru.nl) and Elise D. Bree (Res. Inst. of Child Education and Development, Univ. of Amsterdam, Amsterdam, Netherlands)

Adult aging is frequently accompanied by hearing loss, as well as by cognitive decline. Both play a role in listeners' understanding of noisy, foreign-accented speech and fast conversational speech. Hearing loss leads to increased perceptual effort during listening, which affects memory encoding of the spoken message. Furthermore, some studies have suggested that acquired hearing loss also has long-term effects by degrading the quality or accessibility of phonological representations in long-term memory. This was investigated further by looking into non-word reading, thus bypassing immediate effects of hearing loss. A sample of 29 older adults, varying in degree of high-frequency hearing loss and visual digit span performance, saw 72 multisyllabic nonwords varying in phonotactic frequency (i.e., the phoneme-co-occurrence statistics of the language). They saw the nonwords for 5 seconds and were prompted to produce them from memory after another 3 seconds. As expected, response accuracy was influenced by phonotactic frequency of the nonword and digit span performance. Crucially, response accuracy was also higher if the participant had better hearing, supporting the claim that hearing loss degrades phonological representations in long-term memory. These results emphasize the broad consequences hearing loss has on language processing beyond its immediate effect on speech audibility.

11:25–11:45 Panel Discussion

Session 4aUW

Underwater Acoustics and Signal Processing in Acoustics: Environmental Variability Impact on Shallow Water Acoustics I

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Anthony L. Bonomo, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78713

Chair's Introduction—8:00

Contributed Papers

8:05

4aUW1. A traditional forward scattering theory view of shallow water acoustics. Timothy F. Duda (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Theories of propagation through a random medium (forward scattering theories) identify the scales of feature structure within the medium that are most effective at creating field fluctuations. These scales are near the Fresnel scale for straight-line and refracted propagation. The Fresnel scale (or Fresnel radius), the square root of the wavelength times the propagated distance, is therefore useful as a diagnostic. Shallow-water propagation of sound below 1 kHz often shows a dispersed modal propagation character, with modes having distinct wavelengths. Consideration of individual modes reduces the problem physical dimension by one. Waveguide anomaly features that are known to cause fluctuations, such as small-scale internal waves and frontal intrusions, have characteristic scales that overlap with the Fresnel scales for frequencies in the range of 0.1 to 1 kHz and distances of 1.0 to 25 km (40 to 600 m). This means that established theories can be applied, as long as other requirements are met. The waveguide thickness (water depth) can also exhibit variability on these scales, causing a similar fluctuation effect, although in this case the fluctuation represents a departure from acoustic behavior with a simple planar seabed model. This scaling analysis is potentially most useful for modeling shallow-water ocean propagation, or for processing ocean data to emphasize features relevant to acoustic scattering.

8:20

4aUW2. Forward and back scattering in a shallow inhomogeneous environment. Steven A. Stotts, David P. Knobles, and Robert A. Koch (Environ. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd, Austin, TX 78759, stotts@arlut.utexas.edu)

An experiment in 20 m of water off Panama City, FL, motivates the simulation with several propagation models of the time series produced from rough seabed scattering. Both the forward and backward scattered time series are simulated on vertical line arrays in the 2–4 kHz band to ranges out to 5 km. Results from several Born approximation approaches are compared to the results of a full numerical two-way coupled-mode solution to analyze the measured data. The results from a method of separating the forward and backward scattered amplitudes are also presented. Competing effects, such as the interplay between modal attenuation and the scattering of the higher order modes, are examined within the context of both the full coupled-mode and Born approximation solutions.

8:35

4aUW3. Time-domain Helmholtz-Kirchhoff integral for forward surface scattering in a refractive medium. Youngmin Choo, Heechun Song (Scripps Inst. of Oceanogr., 9263 Regents Rd., La Jolla, San Diego, CA 92037, ymchu@ucsd.edu), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Time-domain Helmholtz-Kirchhoff integral (H-K integral) for forward surface scattering is extended to a refractive medium. Ray theory is applied to obtain the Green's function, while the normal derivative in the integral is evaluated analytically using a ray geometry along with the ray-based Green's function. This approach allows for stationary phase approximation which reduces the surface integration to a line integration. For high-frequency signals with a narrow bandwidth, an asymptotic form of the time-domain H-K integral then can be derived using the Fourier transform. The pressure field scattered from a sinusoidal surface wave in a constant-gradient sound speed profile is evaluated and compared to the result based on a conventional ray model.

8:50

4aUW4. A modeling approach to scattering and reverberation in shallow water. Anatoliy Ivakin (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A physics-based modeling approach is described that allows prediction of reverberation in complex shallow water environments. An integral expression is presented for the backscatter intensity with a factorized integrand comprised of two kernels, the two-way propagator and the scattering kernel. The propagator is defined by Green's function, describes the local intensity, and can be calculated using available models, such as PE, normal modes, or ray approximations. An expression for the scattering kernel is obtained using a unified approach to volume and roughness scattering [Ivakin, JASA Feb. 1998], and represents a sum of volume and roughness scattering coefficients. They are specified for sea-water column and rough heterogeneous seabed with continuous spatial fluctuations of compressibility and density, and/or discrete randomly distributed targets, such as bubbles, fish, shells, and others. Results of numerical simulations for shallow water reverberation time/range series, based on a PE propagation model, are presented, and potential contributions of different mechanisms of scattering are compared and discussed. The approach is applied to consider reverberation in a complex shelly sand/mud environment, such as one at the Target and Reverberation Experiment 2013 (TREX), and results in model/data comparisons based on analysis of TREX acoustic scattering data and environmental ground truth measurements. [Work supported by ONR-OA.]

9:05

4aUW5. Simulating the lateral line with low-frequency nearfield acoustic holography based on a vector hydrophone array for short-range navigation in littoral waters. Tim Ziemer (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, tim.ziemer@uni-hamburg.de)

Fish use the lateral line system to detect swimming objects within a range of a few body lengths to avoid collision. This is achieved by detecting particle velocity and acceleration with receptors which are distributed all over the skin. In this work, this principle is imitated by means of a hydrophone array detecting particle accelerations. In a two-dimensional simulation setup, a dipole source is detected in presence of high-level noise and a disturbing source. This is achieved by a hull mounted vector hydrophone array applying low-frequency nearfield acoustic holography and an adaption of minimum energy method which simulate the lateral line. Noise is added to simulate decorrelated signals—like ambient noise and reverberation—whereas a disturbing dipole source creates highly correlated disturbing signals at the hydrophone positions which is typical for early reflections and sound sources outside the detection area. At the hydrophone array the noise is up to almost 70 dB louder than the radiated source signal. Despite these difficult conditions the proposed method localizes the source reliably within the range of a few array lengths. The system could be implemented in vessels for short-range navigation in coastal and littoral areas or underwater vehicles for mine deployment and harbor construction works.

9:20

4aUW6. Model/data comparisons of high-frequency backscattering from well-characterized sand sediment. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

In the Spring of 2010, acoustic backscattering from an artificially-smoothed sand sediment was measured in the NSWC PCD test pond. The measurements were made at frequencies from 200 to 500 kHz as a function of grazing angle. The residual roughness of the smoothed surface was measured using a laser line scanner while the sediment parameters were determined through diver core analysis and sound speed and attenuation measurements. At the time of the experiment, it was not possible to characterize the volume heterogeneities within the sediment. Subsequent analysis of historic conductivity probe measurements collected in the same test pond under similar conditions has been used to fill this gap. With this additional dataset, the environmental characterization of the sediment places significant constraints on models of scattering from both the sediment roughness and the subsurface volume heterogeneities. Using the sediment parameters, the predictions of small perturbation roughness and volume scattering theories are compared to the data when the sediment is modeled as a fluid and as a poroelastic medium. The effect of the measured porosity fluctuations on sound propagation within the sediment is also examined using a recently developed scattering loss model. [Work supported by ONR.]

9:35

4aUW7. Reverberation modeling approximately accounting for three-dimensional forward scattering effects. Eric I. Thorsos, Jie Yang, and Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

A transport theory approach has been developed for modeling shallow water propagation and reverberation at mid-frequencies with emphasis at 1–3 kHz. With this approach, sea surface forward scattering can be taken into account in a 2-D (range-depth) approximation. While the effects of surface forward scattering on transmission loss are found to be modest (1–2 dB), the corresponding effects on reverberation level for typical conditions can be significant (~10 dB), even though bottom backscatter dominates reverberation in shallow water. An attempt to make data/model comparisons brought out the issue of the proper way of using the 2-D (in horizontal plane) surface roughness spectrum to model forward scattering with a 2-D (range-depth) propagation model. For reverberation modeling, we are investigating the approximation of preserving the distribution in vertical scattering angles in reducing the full 3-D problem to a 2-D (range-depth) problem when using the traditional N ' 2-D approach. A description of the method will be given, followed by data-model comparisons for TREX-13. [Work supported by ONR Ocean Acoustics.]

9:50

4aUW8. Coherent, very high frequency underwater acoustic communications under wind-driven seas: Experiments in an ocean simulator. James Preisig (JPAnalytics LLC, 638 Brick Kiln Rd., Falmouth, MA 02540, jpreisig@jpanalytics.com) and Grant Deane (Scripps Inst. of Oceanogr., La Jolla, CA)

Very high frequency, underwater water acoustic communications (VHF UWAC) operated near the sea surface are subject to interference from surface-scattered energy and, under wind driven seas, noise from breaking waves. Here, we present results from experiments to study these effects conducted in an ocean simulator. Acoustic transmissions in the frequency range (400–750) kHz were made using vertical arrays of transducers in the 44 m long, 2.4 m wide, and 1.5 m deep wind-wave channel at the Hydraulics Laboratory at Scripps Institution of Oceanography. This facility simulates oceanic conditions with a piston wave maker and a fan driving airflows up to 15 m/s. A variety of communications and probe signals were sent under calm and wind-driven conditions. Surface scattering was characterized in terms of arrival delay and gravity wave focusing, along with measurements of breaking wave noise. These data will be discussed in terms of VHF UWAC performance.

10:05–10:20 Break

Invited Papers

10:20

4aUW9. Environmental factors that contribute to high frequency bottom loss variability. Jacob George, David W. Harvey, Allen Lowrie (NP53, NAVOCEANO, 1002 Balch Blvd., Stennis Space Ctr., MS 39522, jacob.george1@navy.mil), and Lori S. Conner (NP64, NAVOCEANO, Stennis Space Ctr., MS)

We discuss three factors: short range sediment variability, bio-attenuation, and seafloor roughness. In a previous presentation (ASA 2013), we have shown that statistical distributions of bottom loss values derived from measured transmission loss (TL) are nearly invariant to measured sediment properties such as sound speed, density, and porosity. To test if this surprising result was caused by under-sampling of sediment cores, three TL runs were done in 2014 when cores were collected at 1 km intervals. The results of these showing short range variations will be discussed. Among the stations in our HFBL database we have found possible evidence for bio-attenuation due to fish (anchovies) swim bladder resonance (Diachok-Wales, JASA 2005). Finite element modeling of a TL station has shown that seafloor roughness can explain the observed frequency dependence of HFBL values (Isakson, report to NAVO). All these will be discussed.

10:40

4aUW10. Finite element modeling of propagation and reverberation shallow water waveguide with a variable environment. Marcia J. Isakson, Nicholas P. Chotiros, and James Piper (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Shallow water waveguides can exhibit environmental variability in the water column, on the bottom interface and in the sediment. Finite element models provide a method to capture the effects of this variability since every element can be described by a different sound speed and density. However, fully three-dimensional finite element models are often computationally inaccessible due to extreme memory requirements. In this study, a longitudinally invariant finite element model is used to predict the reverberation from a shallow water waveguide described by environmental measurements at the Target and Reverberation Experiment 2013 conducted off the coast of the Florida panhandle. Longitudinally invariant models retain all of the fidelity of a three-dimensional model with the requirement that one geometric dimension must be invariant. Therefore, it is an ideal model for wedges and ridges. In this case, the longitudinally invariant direction describes the sand ripples. The reverberation with and without variations in sediment sound speed and density will be compared for the same bathymetry to determine the role of sediment variability in reverberation. Reverberation from along and across the sand ridges will also be examined. [Work supported by ONR, Ocean Acoustics.]

11:00

4aUW11. Initial considerations on effects of braided river beds on long-range acoustic propagation in shallow water. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Allen Lowrie, and Rhett Hamiter (Naval Oceanographic Office, Stennis Space Ctr., MS)

A braided river consists of multiple small channels that divide and recombine numerous times. They form when the sediment load and transporting energies are such that coarser sediment can be deposited as shifting islands or bars between the channels. Braided rivers formed on continental shelves during sea level lowstands during the last 700,000 years when the shelf was sub-aerially exposed to greater than 125 m below present day sea level. A salient feature is that the stream bed (which can be many kilometers wide) is expected to exhibit extremely high geoacoustic variability, given that the islands and bars exhibit much coarser material than that in the channels. The variability is expected both laterally (highest variability or smallest scales perpendicular to the stream flow) and also vertically inasmuch as the river channels experience different avulsion patterns over time. Chirp sonar and core data provide some insight into the underlying geologic processes and the associated scales of variability. Modeling gives some initial clues about the effects of braided river beds on acoustic propagation. [Work supported by the Naval Oceanographic Office and the Office of Naval Research, Ocean Acoustics program.]

11:20

4aUW12. Model-data comparisons of range-dependent shallow-water reverberation including both boundary and volume scattering. Dale D. Ellis (Phys. Dept., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com) and Sean Pecknold (Atlantic Res. Ctr., Defence Res. and Development Canada, Dartmouth, NS, Canada)

A range-dependent shallow-water reverberation model using adiabatic normal modes has been previously developed [Ellis *et al.*, ISURC Conference, La Spezia, 2008] to handle bottom scattering and clutter echoes in a range-dependent environment. It has now been extended to handle volume scattering from the water column and volume scattering from the sub-bottom. Beam time series similar to that from a horizontal line array can be produced. Comparisons can then be made directly with data, and area scattering maps created. Of particular interest will be data obtained on the triplet line array during the 2013 TREX experiment in the Gulf of Mexico off Panama City, Florida. Predictions will be compared with data for average reverberation from a number of pings, as well as ping-to-ping variability on a particular run. [Work supported in part by U.S. Office of Naval Research, Code 32.]

Session 4pAAa

Architectural Acoustics and Noise: Acoustic Comfort in Building Indoor Environmental Quality (IEQ) Performance II

Kenneth P. Roy, Cochair
Building Products Technology Lab, Armstrong World Industries, 2500 Columbia Ave, Lancaster, PA 17603

Donna A. Ellis, Cochair
The Division of Architecture and Engineering, The Social Security Administration, 415 Riggs Ave., Severna Park, MD 21146

Invited Papers

1:00

4pAAa1. Annoyance perception of complex multi-tone noise signals in both harmonic and inharmonic structures within the built environment. Joonhee Lee and Lily M. Wang (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Assorted building mechanical systems generate tonal components within the background noise of built environments. In most cases, this type of noise includes multiple tones in harmonic or inharmonic structures rather than a single tone. However, there is limited information on the comprehensive annoyance caused by multiple tones as perceived by human occupants. Two current standards, ISO 1996-2 and ANSI S1.13, propose calculation methods to address tones in noise, but those methods only analyze the tones individually. This paper aims to investigate how each tone contributes to overall annoyance perception when complex tones are present in background noise. Noise stimuli with five-tone complexes between 125 Hz to 2 kHz were artificially generated for subjective testing. The levels of each tone were randomly adjusted for every trial, and both harmonic and inharmonic structured tone complexes were utilized. Ten musically trained subjects participated in the subjective test involving paired comparisons. Each participant was asked to choose which noise stimulus is more annoying between two noise signals. Perceptual weighting analysis is applied to the results to compute a spectral weighting function for overall annoyance. The performance of the derived spectral weighting function is examined against annoyance ratings of actual building mechanical noises.

1:20

4pAAa2. Innovative approaches to structure borne vibration reduction. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

People feel vibrations depending mostly on the vibration direction. One has to distinguish between vertical vibration and horizontal vibration, the latter often called lateral vibration, in order to address the homeowners' concerns. The difference in perception is described in ISO 8041:2005. Above 20 Hz and up to 1000 Hz, vibration may be called structure-borne noise. Such vibrations may generate audible sound. In private or multi-family residence, structure-borne noise can become prevalent due to elevators, mechanical equipment, subways, nearby train tracks, etc. Often times, areas exhibiting these problems do not have the space in floor or wall configurations in order to take a traditional approach toward the abatement of this structure-borne noise. The purpose of this paper is to present the results from a controlled experiment of different combinations of floor and wall configurations under both horizontal and lateral vibration in order to simulate vibration found in the field due causing structure-borne noise. Floor and/or wall configurations will include combinations of various resilient layers in order to present configurations to successfully abate vibrations originating from these sources while conserving and minimizing space in floor and/or wall configurations to optimize the acoustic efficacy of the installation.

1:40

4pAAa3. Acoustic comfort in closed rooms often means expectations of speech privacy. Kenneth W. Good and Kenneth P. Roy (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

The design of privacy areas is very similar in offices, healthcare, and other building segments, but the focus is squarely on healthcare due to the HIPAA regulations involving personal medical information. Given the degree of enclosure provided by the walls, ceiling, doors, etc., common expectations by occupants of those closed spaces is for confidential speech privacy. So, how do we design and evaluate the speech privacy provided by medical offices, treatment rooms, conference rooms, etc. ASTM laboratory testing of components and systems including STC, CAC, and NRC are part of the answer. But the bottom line is in the field with actual performance testing of the systems as they are built and operated. Two questions to be answered are how do we make informed decisions about product and system choices, and how do we maximize speech privacy using those architectural choices.

Contributed Papers

2:00

4pAAa4. Acoustic comfort in living environment. Made Samantha Wiratha and Lucky Tsaih (Architecture, National Taiwan Univ. of Sci. and Technol., RB 807, No.43, Sec. 4, Keelung Rd., Da'an Dist., Taipei City 10607, Taiwan, sasawiratha@gmail.com)

As population of aging people growing fast, the need of long-term care facility is also increased. In the living environment of elder people, acoustic comfort has the same importance as thermal comfort and visual comfort. This research is to study the preference of acoustic comfort through the listening evaluation with normal hearing people. The listening evaluation is based on 20 live recorded sound samples from two Taiwanese long term care facilities and a university dormitory. The listening evaluation was participated by 66 architecture students. A semantic differential scale questionnaire with 11 pairs of sound qualities for each sound sample were used. The results shown that normal hearing people have negative impression for the current acoustic comfort condition of these Taiwanese long term care facilities. All of the respondents gave negative impression to water pump machine sound due to the "noisy" and "harsh" sound qualities. A live percussive music rehearsed event was associated with "noisy" and "agitating" impression by 97% of the respondents. Speech, TV, telephone ringing, snoring, and footsteps were also identified as discomfort sound quality. Bird chirping and quite dormitory room were recognized by 91% of the respondents as the most comfortable sound with the "natural" and "quiet" impression.

2:15

4pAAa5. Evaluation of a real-time convolution system for perception of self-generated speech in simulated rooms. Jennifer K. Whiting, Timothy W. Leishman, Nathan G. Eyring, Mark L. Berardi, and Michael K. Rollins (Dept. of Phys. and Astronomy, Brigham Young Univ., C110 ESC, Provo, UT 84606, lundjenny@comcast.net)

A real-time convolution system has been developed to quickly manipulate the auditory experiences of human subjects. The system is used to study perceptions of self-generated speech and music, and responses of talkers and musicians to varying acoustical conditions. It allows talkers in an anechoic environment to experience simulated room responses excited by their own voices. While their direct sound travels directly to their ears, they hear convolved room responses via specialized headphones spaced away from their heads. This presentation discusses the system's development, as well as its objective and subjective validations. Several existing rooms were modeled using EASE. Oral-binaural room impulse responses (OBRIRs) from these models were generated and implemented with the convolution system. Binaural recordings and measurements from the rooms were also made using a G.R.A.S. KEMAR mannequin. Objective comparisons of the OBRIRs from the measurements and simulations were explored in the investigation. Subjective evaluations of auralizations made from the OBRIR measurements and simulations, and binaural recordings, followed from A/B listening and speaking tests. In the latter, participants spoke in the various simulated acoustical environments and compared and rated the effects of each experience.

2:30

4pAAa6. Effects of noise flanking paths on ceiling attenuation class ratings of ceiling systems and inter-room speech privacy. Gary Madaras (ROCKFON, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com) and Andrew E. Heuer (NGC Testing Services, Buffalo, NY)

Continuous plenums above suspended, modular ceilings and partial-height walls in buildings can result in inter-room speech privacy and annoyance problems, especially when noise flanking paths via air diffusers, grilles, and lights exist. However, testing of the effects of ceiling system noise flanking paths is limited in the industry. Multiple ceiling systems comprised of various noise flanking paths through air diffusers, grilles, and lights were tested in an independent, accredited, acoustics laboratory according to ASTM International (ASTM) E-1414 and E-413. Additionally, recorded speech was played back in the test chamber source room and binaurally recorded in the test chamber receiver room. The results show that wideband ceiling attenuation class (CAC) decreases by 10 decibels (dB) and 1/3 octave band normalized ceiling attenuation (Dn,c) decreases by 15 to 22 dB in the higher frequency bands when common noise flanking paths are introduced into a ceiling system with CAC-37 ceiling panels. Subjective listening during the course of these tests shows that a ceiling system comprised of CAC-37 panels and typical noise flanking paths (that drop the system rating down to CAC-27) did not provide speech privacy. Intelligibility of recorded speech transmitting into the receiver room was high.

2:45

4pAAa7. A case study investigation of the indoor environmental noise in four urban South African hospitals. Coralie A. van Reenen (Built Environment, Council for Sci. and Industrial Res., PO Box 395, Pretoira, Gauteng 0001, South Africa, cvreenen@csir.co.za)

This multiple case study was designed to investigate acoustics in multi-bed general wards in four South African hospitals. Evidence-based research shows that a quiet indoor environment has positive outcomes for hospital patients and staff. Though international guidelines define noise limits in hospitals, numerous studies world-wide reveal that few hospitals, if any, comply. The goal of this research was to determine whether hospital design paradigms in South Africa should be changed to improve the acoustic environment based on the findings of an acoustic assessment. The acoustic conditions in wards were assessed in terms of sound levels, user opinions, and architecture. The objectives were to determine whether the sites comply with guidelines, to determine user perceptions of noise, and to determine whether design factors influence the noise. It was found that the average sound levels exceeded the guidelines, yet the overall user perception was that noise is not disturbing. Layout and workflow have a likely influence on noise, requiring further research with particular reference to the difference found between patient and staff perceptions of sound. Further discussion pertains to the interpretation, application, and relevance of noise guidelines in an operational hospital environment, recommending further extensive research of human responses to noise exposure.

Session 4pAAb

Architectural Acoustics: Architectural Acoustics Potpourri

Ana M. Jaramillo, Chair
Ahnert Feistel Media Group, 8717 Humboldt ave. N, Brooklyn Park, MN 55444

Contributed Papers

3:20

4pAAb1. The status of classroom acoustics in Colombia. Ana M. Jaramillo (AFMG Services North America, LLC, 8717 Humboldt ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu), Bruce C. Olson, and Milton A. Salcedo (Olson Sound Design, LLC, Brooklyn Park, MN)

With the lack of standards to regulate classrooms in Colombia, we decided to find out what the current status was in terms of acoustical comfort. Due to the mild weather, classrooms usually do not have mechanical heating or cooling. For this reason, noise levels are mandated by proximity to busy roads or other noise sources. Their construction is typically done using heavy materials such as brick and concrete, which results in very low noise transmission, but Reverberation Times tend to be higher than recommended due to the little use of absorptive materials. Measurements and visual inspections were performed in several schools to evaluate these parameters against the current S12.60 ANSI standard.

3:35

4pAAb2. Acoustic response of a multipurpose auditorium at Grand Valley State University. Bailey Groendyke and Karen Gipson (Phys. and Music, Grand Valley State Univ., Allendale, MI 49401, groendba@mail.gvsu.edu)

Like many multipurpose auditoriums, the Louis Armstrong Theatre (LAT) at Grand Valley State University has been reported by a considerable number of students and faculty to have unsatisfactory acoustics for music performance. This study focused on physical measurements and simulated changes to LAT. Reverberation time (RT) was measured by filling LAT with sound and measuring the decay for select frequencies as per ASTM E2235 protocol, and the initial time delay gap (ITDG) was determined using slapsticks as an impulsive sound source. A model of LAT was also constructed from blueprints and physical measurements; simulations using this model were conducted using Odeon. Data from the physical measurements as well as the simulation confirmed that the RTs over a wide range of frequencies were smaller than desired for music, whereas ITDG measurements showed prevalent spurious reflections. Modifications to the model were made to increase reverberation time and reduce undesirable reflections in order to improve LAT for musical performance without compromising its functionality for speech.

3:50

4pAAb3. Acoustical characterization of touristic caves in Portugal. Antonio P. Carvalho and Joana I. Sousa (Lab. of Acoust., Univ. of Porto, FEUP (NIF501413197), R. Dr. Roberto Frias, Porto 4200-465, Portugal, carvalho@fe.up.pt)

Since the Palaeolithic, Mankind has been taking advantage of the acoustical characteristics of natural caves to perform its rituals. Nowadays, many of these spaces are used as touristic attractions or even as stages for musical performances. This study characterizes three touristic caves in Portugal where *in situ* measurements were done, of background noise sound pressure levels, RASTI, and Reverberation Time. The average RT values were 1.3 s to 1.7 s, and the RASTI average values revealed good intelligibility (from 0.50 to 0.57). The sound absorption coefficient of the stone that constitutes the interior of those caves was also measured in a standing wave apparatus.

4:05

4pAAb4. Acoustical study of historical large room for a contemporary use: The oratory of Albergo dei Poveri in Genova. Anna Chiari (DSA, Università Degli Studi di Genova, Genova, Italy), Ilaria Pittaluga, Corrado Schenone, and Davide Borelli (DIME - Sez. TEC, Università Degli Studi di Genova, Via all'Opera Pia 15/A, Genova, Italy, corrado.schenone@unige.it)

The conversion of deconsecrated religious buildings in conference halls or music auditoria is nowadays frequent, especially throughout Italy and Europe, although in most cases these rooms are under artistic and architectural preservation due to their historical heritage value. In this way, acoustic control interventions in such spaces, generally characterized by high reverberation time values, tend to be highly complex. The Oratory of "Albergo dei Poveri" building in Genova, transformed during the 90s in the Auditorium of the Departments of Law and Political Sciences of the University of Genova, faced these needs. After several interventions regarding the installation of sound systems and acoustical curtains, nowadays the acoustical features of the room are not still sufficient for the achievement of the adequate acoustic comfort. In this paper, the acoustic analysis of the hall has been evaluated and effective correction interventions at the same time compliant and non-invasive with the architectural constraints have been designed and investigated. A numerical model of the room was implemented through computer simulations and the model was validated by *in situ* measurement campaigns. Then, the numerical model was used to develop the subsequent design-scenarios calculations.

4:20

4pAAb5. A study on the analysis of hanok (the Korean traditional houses) acoustic characteristics. Won-Hee Lee, Myung-Sook Kim, and Myung-Jin Bae (Hyungnam Eng. Buliding 1212, Soongsil Univ., 369 Sangdo-Ro, Dongjak-Gu, Seoul, Korea, kimm@ssu.ac.kr)

Hanok is a type of Korean traditional houses, which is built based on science. In order to build Hanok, necessary materials are wood, rock, and clay, which can be easily acquired in nature. Due to the natural building materials, it is known that hanok is efficient to keep it warm and cool. Its heating system, Ondol, uses thermal conduction and convection of heat from the fireplace. Doors and windows are located on south and east sides because cold wind comes from northwest side in winter and cool breeze blows from southeast side in summer. On the other hand, the genre of music that goes well with the Hanok is the unique melody of Gukak (Korean classic music). The floors and walls of the Hanok, the rafters, highlights Gukak's uniqueness and plays the role of a sound box. This study was conducted to analyze the harmony between Hanok features and Gukak's acoustic characteristics.

4:35

4pAAb6. Assessing the range of spatial impression metrics from varying source position across a stage in assorted performing arts venues. Sungbeen Cho and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, sungbeen@huskers.unl.edu)

A listener's perception of sound in the built environment is determined by various physical factors such as the size and form of a space, and shape and finishing of materials. Spatial impression is a term used to describe the spatial perception of the auditory field by listeners, and a number of metrics have been proposed to quantify this, including Interaural Cross-Correlation Coefficient (IACC) and Lateral Energy Fraction (LEF). While values for these metrics have been presented for a number of performing arts venues, few investigations have studied the range of these metrics within a specific venue, particularly as a source moves across the stage. This paper assesses data from both computer-simulated and measured rooms on how IACC, LEF, and Interaural Level Differences (ILD) vary within a venue, due to early reflections, as the source location varies across the stage. How greatly do the ranges of these metrics differ with source location? The range and values of these spatial impression metrics due to source movement may be another way in which a venue's spatial impression can be quantified and compared, correlating to how a listener at a fixed location spatially perceives sound sources within that room.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — CITY TERRACE 9, 1:00 P.M. TO 5:35 P.M.

Session 4pAB

Animal Bioacoustics and Acoustical Oceanography: Bioacoustics Research in Latin America

Juliana R. Moron, Cochair
Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Rua Batista de Oliveira 1110 apto 404 B, Juiz de Fora 36010520, Brazil

Marie Trone, Cochair
Math and Science, Valencia College, 1800 Denn John Lane, Kissimee, FL 34744

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. New perspectives in Amazonian bird vocal behavior. Maria Luisa Silva (Instituto de Ciências Biológicas, Universidade Federal do Pará, Rua Augusto Correa, 1, Belém, Pará 66075150, Brazil, silva.malu@uol.com.br)

Neotropical region are known for its huge avian diversity and for the dominance of Sub-oscine or Tyranni passerines species. These species diversity are reflected in vocal communication and we will present here two examples of Brazilian Amazon forest of Tyranni species. Although the Tyranni are known for presenting a stereotyped song compared to Oscine passerines, we have found a species with a complex calls repertoire and other with a distinguished individual variation. The study area is the Ecological Park of Gunma, Santa Bárbara, Pará, Brazil, 50 km north of Belém. We performed playback experiments to define the vocalizations of Rusty-margined flycatcher *Myiozetetes cayanensis*, species that presents a complex repertoire of 15 different vocalizations, including a complex duet song. We also studied *Lipaugus vociferans,* a lekking bird, in which males perform conspicuous vocalizations in aggregations. We have analyzed songs of 19 individuals from six leks, considering the physical parameters (frequency bandwidth and temporal parameters) of the species-specific song. The results showed that it is possible to differentiate the individuals by song. These differences can represent an important function in individual recognition inside or between leks. These results show the necessity to investigate the high diversity of behavior in tropical habitats.

1:25

4pAB2. Does vocal learning accelerate acoustic diversification? Evolution of contact calls in Neotropical parrots. Marcelo Araya-Salas, Angela Medina-Garcia, and Timotthy Wright (Biology, New Mexico State Univ., 625 E University Ave., Apt. 26, Las Cruces, NM 88005, maraya@nmsu.edu)

Learning has been traditionally though to accelerate the evolutionary change of behavioral traits. We evaluated the evolutionary rate of learned vocalizations and the interplay of morphology and ecology in the evolution of these signals. We examined contact calls of 51 species of Neotropical parrots of the tribe Arini from recordings obtained in Central and South America. Parrots are ideal subjects due to their wide range of body size and habitats and their open-ended vocal learning that allows them to modify their calls throughout life. We estimated the evolutionary rate of acoustic parameters of parrot contact calls and directly compared them to those of morphological traits

and habitat. We also evaluated the effect of body mass, bill length and vegetation density on acoustic parameters of contact calls while controlling for phylogeny. Evolutionary rates of acoustic parameters did not differ from those of our predictor variables except for spectral entropy, which had a significantly slower rate of evolution. We found support for correlated evolution of call duration, and fundamental and peak frequencies with body mass; and of fundamental frequency with bill length. We demonstrate that parrot contact calls, which are learned acoustic signals, show similar evolutionary rates to morphological traits. This is the first study to our knowledge to provide evidence that change through cultural evolution does not necessarily accelerate the evolutionary rate of traits acquired through life-long learning.

1:45

4pAB3. Transmission properties of vocalizations in a year-round territorial bird. Luis Sandoval (Escuela de Biologia, Universidad de Costa Rica, Escuela de Biologia, Universidad de Costa Rica, Montes de Oca, San Jose 11501-2060, Costa Rica, biosandoval@hotmail.com)

Acoustic Adaptation Hypothesis (AAH) predicts that acoustic signals using in long-distance communication should be optimized for transmission through its natural environment. To test if White-eared Ground-sparrows (*Melozone leucotis*) vocalizations are adapted to transmit long distances, I conducted two sound transmission experiments where I broadcast and re-recorded different calls, songs, and duets. This ground-sparrows use vocal communication year round for territory defense and mate attraction. I conducted the experiments inside ground-sparrows territories in Costa Rica, broadcasting natural vocalizations at different combination of distances, speaker and microphone heights to quantified the signal-to-noise ratio, tail-to-signal ratio, blur ratio, and excess attenuation. Songs and duets of White-eared Ground-sparrows showed similar patterns of degradation with distance and with proximity to the ground, suggesting that vocalizations facilitate communication with receivers at similar shorter distances (in less than a typical territory's diameter). *Chip* calls showed higher degradation in comparison to *tseet* calls with the distance, suggesting that *tseet* calls are design for longer distance communication. To my surprise, *chip* calls, songs, and duets has not experienced strong selection for long distance communication, because results do not support the AHH, and probably these vocalization characteristics are under other selective forces as sexual selection or phylogenetic constrain.

2:05

4pAB4. Studying dolphin whistles in Mexico. Carmen Bazúa-Durán, Julieta E. Sarmiento-Ponce, Brenda P. González-Leal, and Elena Montejano-Zea (Física, UNAM, Facultad de Ciencias, Circuito Exterior s/n, Ciudad Universitaria, México, D.F. 04510, Mexico, bazua@unam.mx)

Dolphin whistles are emitted especially during social interactions and feeding activities involving group cohesion, individual recognition, and recruitment. Here, at the Faculty of Sciences of UNAM we are studying dolphin whistles, mainly those of wild and captive bottlenose dolphins, *Tursiops truncatus*, to learn about their social structure and how whistles may be used to study dolphin ecology. We have developed a new methodology to describe and compare the whistle repertoire, which consists of whistle contour extraction to classify whistles into whistle types (using Matlab BELUGA and ArtWARP), then classifying whistle types into four general categories (high complexity, low complexity, linear long, and linear short), and finally computing a complexity index and a proportional variability of the whistle repertoire. Results obtained showed that this very simple method is useful to describe the whistle repertoire and to compare it according to the general behavioral state of dolphins, and between species. It is necessary to implement new methodologies like this one to better understand how dolphins are using whistles, since acoustic communication is the most important sense in dolphin species. [Work supported by PAPIIT-UNAM.]

2:25

4pAB5. Cetacean acoustic survey using towed array in the western South Atlantic shelf break. Artur Andriolo, Franciele R. Castro, Thiago Amorim (Laboratório de Ecologia Comportamental e Bioacústica, Departamento de Zoologia, Universidade Federal de Juiz de Fora, ICB, Rua José Lourenço Kelmer, S/n - Martelos, Juiz de Fora 36036-330, Brazil, artur.andriolo@ufjf.edu.br), Eduardo R. Secchi, Juliana Di Tulio (Instituto de Oceanografia, Universidade Federal de Rio Grande, Rio Grande, Brazil), Juliana Moron, Gabriela Ramos, Bruna Ribeiro (Laboratório de Ecologia Comportamental e Bioacústica, Departamento de Zoologia, Universidade Federal de Juiz de Fora, Juiz de Fora, Brazil), Alexandre N. Zerbini (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., NOAA, Seattle, WA), Luciano Dalla Rosa (Instituto de Oceanografia, Universidade Federal de Rio Grande, Rio Grande, Brazil), Raíssa R. Mendes, and Fábio B. Palácio (Laboratório de Ecologia Comportamental e Bioacústica, Departamento de Zoologia, Universidade Federal de Juiz de Fora, Juiz de Fora, Brazil)

Passive acoustic towed array during ship surveys has being applied to increase the knowledge about cetacean. From 2012 to 2015, towed arrays were used to investigate cetacean distribution along the western South Atlantic shelf break. Research cruises were performed between 26°S and 38°S over the continental shelf break and slope. Acoustic tracklines comprised an average of 780 nm of effort per survey. Hydrophone arrays (Auset®) were towed 150 and 300 m behind the vessel. The system was configured to give a variable frequency response from 1592 (High Pass Filter) to 100,000 Hz. Acoustic data were recorded as .wav files. Concurrent environmental and GPS data were logged automatically using WinCruz software. Visual positive identifications were associated to the acoustic recordings. The .wav files were analyzed using partially automated detections tools complemented with visual and acoustical searched for species confirmation whenever possible. A total of nine cetacean species were acoustically detected. The most frequently species were *Physeter macrocephalus*, *Delphinus delphis*, *Stenella longirostris*, *Orcinus orca*, and *Globicephala sp.*. Additional studies are needed to describe acoustic parameters of the various species present in this region in order to improve automated detection systems. [This study was supported by Instituto Aqualie and was funded by BG Group and Chevron Brasil Upstream Frade LTDA.]

Contributed Paper

2:45

4pAB6. Enhanced feature extraction using the Morlet transform on 1 MHz recordings reveals the complex nature of Amazon River dolphin (*Inia geoffrensis*) clicks. Marie Trone (Math and Sci., Valencia College, 1800 Denn John Ln., Kissimee, FL 34744, mtronedolphin@yahoo.com), Hervé Glotin (Comput. Sci., LSIS UMR 7296, DYNI, Aix Marseille Université, CNRS, ENSAM, Université de Toulon, Institut Universitaire de France (IUF), La Garde, France), Randall Balestriero (Comput. Sci., LSIS UMR 7296, DYNI, Université de Toulon, La Garde, France), and David E. Bonnett (LCDR USN (Ret), Silverdale, WA)

The Amazon River dolphin lives exclusively in freshwater throughout the Amazon River watershed, a dynamic and acoustically complex habitat. Although generally considered a relatively non-vocal species, recent evidence suggests that these animals are acoustically active, producing tremendous quantities of high-frequency, pulsed signals. Moreover, these pulsed signals appear to be considerably more complex than previously believed. This study explored the high-frequency pulsed emanations produced by Amazon River dolphins in Peru. Audio recordings were made using a two hydrophone array, one of which was sampled at 1 MHz, in August of 2015. Digitized recordings were analyzed using FFT and Morlet wavelets. Subsequently, unsupervised machine learning attempted to delineate various click categories based upon inter-click intervals, the frequency bandwidth of each click, and the formants contained within each click. Although the Morlet transform is much more robust and accurate for higher frequencies than the FFT, its performance was not constant for all frequencies. Thus, the Morlet transform and the FFT produced different click categories. Thus, formant results above 230 kHz most likely were skewed. These results are the first to clearly demonstrate the heterogeneity of the high-frequency pulsed emanations of the Amazon River dolphin.

Invited Paper

3:00

4pAB7. Steps towards promoting bioacoustics research in Mexico. Eduardo Vivas (CIBNOR, Av Instituto Politecnico Nacional 195, Playa Palo de Santa Rita Sur, La Paz, Baja California Sur 23096, Mexico, evivas@cibnor.mx)

Bioacoustics has played a major role in animal research in the past decade. Unfortunately, in Mexico, it is still seen as a novel approach, and few research groups incorporate it into their studies. To overcome this lag, work needs to be done in three major areas: Collaboration among experts, preparing young students, and providing specialized equipment. In Mexico, there are many institutions devoted to research in biology, but there are only a few researchers in acoustics because there are no graduate research programs in this area. Collaboration among acousticians interested in biology and biologists interested in acoustics becomes crucial, and future researchers, coming from both specialties, need to be prepared. Once this synergy is created, emerging research groups often see the cost and availability of specialized equipment as a limitation, so it is also necessary to focus on providing access to research-grade, low-cost hardware. In this talk, I will describe my efforts toward promoting bioacoustics research in Mexico through collaboration with other research centers, with examples of ongoing projects, graduate program support, and equipment presented in major peer-reviewed conferences that we have developed with low-cost, open source materials that are comparable with standard, high quality research equipment.

3:20–3:35 Break

Contributed Papers

3:35

4pAB8. Preliminary evidence for signature and copied whistles among spinner dolphins in the Southwest Atlantic Ocean: Beacon purpose? Juliana R. Moron (Instituto Aqualie, Universidade Federal de Juiz de Fora, Juiz de Fora 36010520, Brazil, julianamoron@hotmail.com) and Artur Andriolo (Laboratório de Ecologia Comportamental e Bioacústica, Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Juiz de Fora, Brazil)

In order to evaluate strategies of cohesion in a fission-fusion society, the occurrence of signature and copied whistles were investigated in free-ranging spinner dolphins, *Stenella longirostris*. Through an one-element hydrophone array towed over the Brazilian continental shelf break, a group of approximate 400 dolphins were recorded at 96 kHz/24 bits while navigating. The preliminary results demonstrated 218 similar signals that fit into previous definitions of signature or copied whistles. These whistles were produced in bouts with an inter-whistle interval of 0.066—11.56 s (mean $\pm$ SD: 2.66 $\pm$ 2.72) that varied from 2 to 32 repetitions comprising six different contour shapes. Thus, these data support previous hypothesis that these signals are important units in the dolphin's repertoire. It may also suggests a potential use on individual direction and localization, where repeated contours could be acting as a beacon to direct and locate the animals within the group. Additional research to ascertain the natural function of these vocalizations may clarify the basis for acoustic badges of membership and group organization of this cosmopolitan species.

3:50

4pAB9. Geographic variation assessment of Bryde's whale (*Balaenoptera edeni*) Be4 call in the Gulf of California. Eduardo Vivas (CIBNOR, La Paz, Baja California Sur, Mexico), Violeta C. Vera Cuevas (CIBNOR, Centro de Investigaciones Biológicas del Noroeste, S.C., Av. IPN 195. Col Playa Palo de Santa Rita Sur, La Paz 23096, Mexico, cassandra_vera@hotmail.com), Jorge Urbán Ramírez, Lorena Viloria Gómora (Programa de Investigación de Mamíferos Marinos. Departamento de Biología Marina, Universidad Autónoma de Baja California Sur, La Paz, Mexico), and Patricia Cortés Calva (CIBNOR, La Paz, Mexico)

Mysticeti whales modify their calls in response to an increase in noise level within their environment. This is particularly important since environmental noise has been increasing continuously in the last decades. In the case of Bryde's whale (*Balaenoptera edeni*), previous studies of its predominant call, Be4, have exposed differences in the duration of its main component (f0) for northern and southern areas of the Gulf of California (GC). The objective of this research is to determine if the differences found among Be4 calls (f0, duration, and energy distribution) are influenced by the noise levels in those areas. Results based on the characterization of Be4 Call and noise show that northern noise level is intense in the octave frequency band centered around 30 Hz, and could mask the 30 Hz component. This kind of noise is rare in the south (5%) and when it was present, the 30 Hz component was not registered, which might be an adaptation to noise. Predominant

moderate noise in the south within the band 20 to 70 Hz overlaps the signals and higher components are present, especially the 165 Hz component. Results suggest that the difference between call from north and south areas might also be related to the noise present.

4:05

4pAB10. An underwater acoustic camera for marine mammal vocalization interaction studies. Eduardo Vivas (CIBNOR, La Paz, Baja California Sur, Mexico), Omar Bustamante, and Sergio Beristain (Academia de Acustica, ESIME, ESIME Academia de Acustica, Instituto Politécnico Nacional, Instituto Politécnico Nacional s/n, Unidad Profesional Adolfo Lopez Mateos, Gustavo A. Madero, Mexico City, Distrito Federal 07738, Mexico, omarb.p@hotmail.com)

Acoustic Cams, promoted as "a way to listening with your eyes," are widely used in the industry to visually pinpoint particular sound sources in a clouded sound environment. A video camera and complex multiple microphone array configurations are used to find the direction of arrival of sound, and the results are color coded and superimposed in the video image. This technique has a great potential to be used in marine mammal studies where underwater vocal interactions among a close group needs to be determined. This is particularly important for *Otariids* and *Pinnipeds* for which the function of in-water calls involving no bubble emission needs to be studied. A prototype of an underwater acoustic camera built around a low cost open source linear array of hydrophones and a fishing underwater camera is presented. Results of the prototype test, first under controlled conditions, and then recording vocal interactions during normal swimming behavior of a colony sea lions (*Zalophus californianus*) in the Gulf of California, are presented.

4:20

4pAB11. Bioacoustic characterization of a coastal marine soundscape in Quintana Roo, Mexico. Heather R. Spence (City Univ. of New York Graduate Ctr., Psych. Dept., Hunter College, 695 Park Ave., New York, NY 10065, info@heatherspence.net) and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Wild dolphins are exposed to a variety of biological and other natural and anthropogenic sounds. Several species of dolphins, including bottlenose dolphins (*Tursiops truncatus*), have been documented in the MesoAmerican Reef (MAR) region; however, there is little to no scientific monitoring of these populations. Characterizing the soundscape of wild dolphins on a section of the MAR would facilitate understanding the acoustic nature of this dolphin habitat. Passive Acoustic Monitoring using an Ecological Acoustic Recorder was conducted for one year off the coast of Quintana Roo, Mexico, just north of Isla Mujeres, a site where dolphins are known to be frequent. The soundscape was typified by natural sounds, however there was important periodic contribution by anthropogenic sources. Boat motor noise contributed to higher SPL_{rms} during the day than in the night and contributed to noise between 500 Hz and 25 kHz, which is consistent with small vessels. While boat motor noise was not as frequent as fish sounds nor as pervasive as snapping shrimp sounds, when present it potentially overwhelms the natural soundscape. Anthropogenic characteristics of this soundscape have implications for dolphin welfare and regulations in the nearby marine protected areas.

Invited Papers

4:35

4pAB12. Assessing the structure of a Neotropical bat community using acoustic monitoring techniques. Sergio Estrada Villegas (Smithsonian Tropical Res. Inst., Dept. of Biological Sci., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, estradavillegassergio@gmail.com), Christoph F. Meyer (Universidade de Lisboa, Lisboa, Portugal), Brian McGill (Univ. of Maine, Orono, ME), and Elisabeth K. Kalko (Univ. of Ulm, Ulm, Germany)

Determining the structure and composition of tropical communities is challenging because some species are rare or hard to detect. Within Neotropical bats, aerial insectivores have been systematically undersampled because they avoid mist nets, the traditional sampling tool. Advances in bioacoustic monitoring techniques have allowed the study aerial insectivorous bat (AIB) communities across various spatial scales and habitats. We present two studies that assessed the underlying mechanisms that structure an AIB community across the Isthmus of Panama. First, we evaluated how habitat fragmentation affected two guilds of AIBs and found higher species richness in islands than in continuous forests. Background clutter aerial insectivores showed compositional differences due to effects of isolation, area, and forest complexity, whereas open space bats were not affected by fragmentation. Second, we determined how climate and forest complexity affected AIB community structure at different spatial scales. We found that most of the variation in bat richness, abundance, and feeding activity occurred at the smallest spatial scale (10×10m) and was explained by habitat structure. In contrast, at large scales, climatic differences explained most of the variation in individual species' abundances. Interestingly, species richness peaked at intermediate levels of precipitation, while total abundance was very similar across sites.

4:55

4pAB13. Acoustic description and synchronization of the duetting species *Pezopetes capitalis*, Costa Rica. Carla V. Trejos (Biology, Universidad de Costa Rica, Santa Clara, San Carlos, Santa Clara, Alajuela 21001, Costa Rica, ktrejos07@gmail.com) and Gilbert Barrantes (Biology, Universidad de Costa Rica, Tres Rios, Cartago, Costa Rica)

Duetting has been widely studied, and its function vary across species, though in all cases temporal synchrony of the elements sang by each sex in the duet seems to play an important role. My goals in this study were to describe the structure of the duet of *Pezopetes capitalis*, and the degree of temporal synchronization of each individual of the pair during duetting. Duets of *P. capitalis* consist of overlapping elements between both members of a mated pair and can be initiated by either sex. Synchronization was similar between both individuals. For 11 pairs, both, females and males, reduce their silence intervals before singing a new element as a response to an increase in the duration of its partner's elements. Furthermore, both mates lower the high frequency of their elements of the second section, as the duet continues. Birds showed this pattern even in those cases where the female joins the male after he had began to sing its duet's part. This suggests a temporal coordination in frequencies (high frequency decreases with time). Indirect evidence suggest that duets in *P. capitalis* serve as a joint defense of territory and as a way to recognized mates when a pair reunited.

5:15

4pAB14. Scaled mining of environmental acoustic data from temperate to tropical forests, from ocean to tropical rivers: A convolutional feature learning approach. Herve Glotin (LSIS UMR 7296, DYNI, Comput. Sci., Aix Marseille Université, CNRS, ENSAM, Université de Toulon, Institut universitaire de France (IUF), USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr)

Scaled passive acoustic monitoring has recently been developed to assess changes in biodiversity. It is applied to monitor changes in fauna composition associated with anthropogenic impacts and to improve the management of species and habitat conservation. Under the auspices of the "Scaled Acoustic BIODiversity" SABIOD project (CNRS MI/GDR MADICS/JASON UTLN), interdisciplinary teams collaborate to develop new joint machine learning and signal processing bioacoustic analyses. Currently, terabytes of sounds are recorded monthly using the innovative open SABIOD autonomous sensor arrays, positioned within forests or deep in the ocean. The challenges associated with tropical biodiversity as the variety of sound sources and the complexity of acoustic paths are being addressed by optimizing strategies that couple features learning, convolutional neural net, and advanced passive acoustic localization/filtering. Applications are many, including the classification of one thousand Amazon bird species, whale song classification, automatic indexing forest soundscapes, passive acoustic 3D tracking of bats and cetaceans with our patented system, and offshore cetacean monitoring. We discuss the main difficulties encountered and summarize the promising steps and strategies that we intend to pursue in the future coupling scaled CNN with Deep Scattering Spectrum. Demo/details: http://sabiod.univ-tln.fr.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — CLEARWATER, 1:30 P.M. TO 4:55 P.M.

Session 4pBA

Biomedical Acoustics and Physical Acoustics: Numerical and Analytical Modeling of Medical Ultrasound II

Martin D. Verweij, Cochair
Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, Lorentzweg 1, Delft 2628CJ, Netherlands

Robert McGough, Cochair
Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824

Chair's Introduction—1:30

Invited Papers

1:35

4pBA1. Simulation of wave propagation in anisotropic and viscoelastic tissues. Matthew W. Urban, Bo Qiang, Sara Aristizabal, Ivan Nenadic, Pengfei Song, Shigao Chen (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Many methods for characterization of the mechanical properties of soft tissues using propagating shear waves have been developed over the past two decades. Most of these methods assume that the shear wave is traveling in an elastic, isotropic tissue. However, many soft tissues are viscoelastic and have material properties that are directionally dependent or anisotropic. We have been developing methods to measure waves propagating in soft tissues to estimate the anisotropic viscoelastic material properties. To refine our measurement methods, we have also developed techniques and models for simulating the wave propagation in these types of materials. We have developed specialized finite element and pseudo-spectral models that can simulate viscoelastic transversely isotropic materials and compared our results with measurements in *ex vivo* porcine muscle. Additionally, we have developed a finite element model that simulates wave propagation in the myocardium by creating a model of layered transverse isotropic media oriented at different angles. We examined the effect of frequency of the waves for estimating of the angle of propagation. We compared these simulation results with experimental results from an *ex vivo* porcine left ventricular wall. These simulation methods provide insight for optimizing our measurement methods for *in vivo* characterization of material properties.

1:55

4pBA2. Numerical simulation of transcranial focused ultrasound therapy. Aki T. Pulkkinen (Dept. of Appl. Phys., Univ. of Eastern Finland, PO Box 1627, Kuopio 70211, Finland, Aki.Pulkkinen@uef.fi), Beat Werner, Ernst Martin (Ctr. for MR-Res., Univ. Children's Hospital, Zürich, Switzerland), and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

Transcranial focused ultrasound therapy is becoming a viable tool for treatment of brain related disorders. The technique has been used for treatment of essential tremors and chronic neuropathic pain with good results. In addition, the technique has a broad scope of potential applications including: tumor ablation, localized drug delivery, thrombolysis, and neurostimulation. Performing an efficient focused ultrasound treatment delivery requires planning. Accurate computational models could, in principle, be utilized to provide auxiliary guidelines for such planning. In this work, a computational model is presented for simulating the propagation of ultrasound in transcranial setting. The effects of ultrasound propagation through soft tissue and bone are modeled by coupling wave equations of fluid and solid. The coupled model is numerically evaluated by using a hybrid simulation technique that couples finite difference method with a grid method. The resulting computational model is utilized to simulate focused ultrasound field in clinical patient treatment setting. Effect of absorption of heat into the brain tissue is further simulated by using the bioheat equation. Simulation results are compared with magnetic resonance thermometry performed during the clinical treatments for evaluation of validity of the computational model.

2:15

4pBA3. Viscosity-compensated shear speed imaging with acoustic radiation force. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824), Matthew W. Urban (Mayo Clinic College of Medicine, Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, mcgough@egr.msu.edu)

Shear wave elastography imaging (SWEI) with ultrasound uses an applied acoustic radiation force to obtain quantitative images of liver, breast, prostate, and other soft tissues. In SWEI, shear velocity waveforms are collected along a line, and then, the shear speed is estimated either with a k-space or a correlation-based method. Although these yield effective estimates in elastic media, both methods consistently overestimate the shear speed in viscoelastic soft tissue because neither considers the shear viscosity. To address this problem, we have created a new shear speed estimation approach that also accounts for the effect of shear viscosity. The new approach is evaluated in a computational model that calculates the three-dimensional (3D) intensity field generated by a linear array transducer in FOCUS (http://www.egr.msu.edu/~fultras-web) and then simulates the shear wave velocities in a viscoelastic medium with a 3D finite difference model. In the viscoelastic shear wave simulation model, the shear speed constant is 1.5 m/s, and the shear viscosity is 1 Pa·s. Correlation-based shear speed estimates are compared to the viscosity-compensated estimates, and the results show that the viscosity-compensated approach consistently achieves improved shear speed estimates relative to the correlation-based method. [Work supported in part by NIH Grant Nos. R01EB012079 and R01DK092255.]

2:35

4pBA4. Simulation of shear wave elasticity imaging including speckle and refraction effects. Stephen A. McAleavey and Jonathan H. Langdon (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

We present an overview of, and results from, a GPU-accelerated, 3D finite-difference time-domain acoustic radiation force impulse (ARFI) shear wave elasticity imaging (SWEI) simulator recently developed in our laboratory. The simulator allows modeling of the ultrasound "push" beam used to generate the shear wave, propagation of the shear wave in a viscoelastic, inhomogeneous medium, and simulation of ultrasound tracking of the shear wave. Spatial variations in both the ultrasound and shear wave speeds can be included to model ultrasound beam refraction errors and shear wave reflection and refraction by inclusions. Speckle has been shown experimentally to induce distortions in the apparent shape of a tracked shear wave, as well as biases in shear wave arrival time estimates used to generate shear wave speed images. A complete simulation of fully developed speckle captures this speckle bias effect but is time consuming. We present a dominant-speckle simulation approach that allows realistic modeling of the speckle-noise induced noise observed in videos of shear wave propagation in distributed speckle targets, and of artifacts in shear wave speed images. Matched simulation and phantom images of uniform and spherical inclusion targets are presented to demonstrate the ability of the simulation to capture key effects.

Contributed Paper

2:55

4pBA5. Numerical simulations of ultrasound-induced deformation of the lung surface. Brandon Patterson and Eric Johnsen (Univ. of Michigan, 1231 Beal Ave., Rm. 2016, Ann Arbor, MI 48109, awesome@umich.edu)

Ultrasound-induced lung hemorrhage remains the only known bioeffect of non-contrast, diagnostic ultrasound (DUS) found to occur in mammals. However, a fundamental understanding of the ultrasound-lung interaction is still lacking. In this study, we numerically simulate the deformation of the lung surface when subjected to clinically relevant pressure waves. We model the lung as air and the surrounding tissue as water. The two-dimensional, compressible Navier-Stokes equations are solved using a high-order accurate finite volume scheme. We first consider an air-water interface with a small sinusoidal perturbation and study the growth of the perturbations when subjected to a planar waveforms consisting of a step change in pressure. Scaling laws relating the waveform properties to the interface deformation rate are proposed for this simple case. Simulations with DUS waveforms indicate that the interface deformation depends on the unsteady nature of the wave. Preliminary analysis suggests that the initial interface deformation is driven by baroclinic vorticity deposited along the perturbed interface, and that the resulting stresses and strains are a possible candidate causing bioeffects.

3:10–3:30 Break

Invited Papers

3:30

4pBA6. Acoustics of finite Bessel tractor beams. Farid G. Mitri (Area 52 Technol., Chevron, 5 Bisbee Court, Santa Fe, NM 87508, f.g. mitri@ieee.org)

Surprises and counterintuitive effects occurring in physical phenomena rank high on the list of things that make science discoveries a source of perpetual excitement. Some examples illustrated here include, **(i)** the effects of attracting (i.e., pulling) a spherical particle placed in the field of a Bessel acoustical beam of progressive waves back towards the radiator's surface [Mitri, "*Single Bessel tractor-beam tweezers*," Wave Motion **51**, no. 6, pp. 986–993, 2014], **(ii)** suppressing some of the sphere's resonances [Mitri, "*Near-field acoustic resonance scattering of a finite Bessel beam by an elastic sphere*," IEEE Trans. Ultrason. Ferroelectr. Freq. Control, **61**, no. 4, pp. 696–704, 2014], **(iii)** and the emergence of a "negative radiation torque," meaning that an incoming high-order Bessel vortex beam with a positive topological charge (known also as the order of the beam) would rotate a viscoelastic sphere in the opposite sense of the beam handedness [Silva *et al*., "*Radiation torque produced by an arbitrary acoustic wave*," Europhys. Lett., **97**, art. no. 54003, 2012]. Results and numerical predictions illustrate the analysis and extensions to other geometries, such as rigid oblate and prolate spheroids [Mitri, "*Axisymmetric scattering of an acoustical Bessel beam by a rigid fixed spheroid,*" http://arxiv.org/abs/1505.06754v3, *ibid*. "*Acoustic radiation force on oblate and prolate spheroids in Bessel beams*," Wave Motion, **57**, pp. 231–238, 2015] are also mentioned.

3:50

4pBA7. Acoustic fields and ultrasound contrast agents. John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Ultrasound contrast agents are encapsulated gas bubbles which oscillate nonlinearly upon acoustic excitation. Recent developments include targeted agents for molecular imaging applications and ultrasound contrast agents such those with polymer shells specifically designed for high frequency ultrasound applications. Though a variety of models have been proposed since the early development of the agents, many outstanding questions exist on the specifications of the shell and most appropriate model for the different materials. Moreover, only recently have independent mechanical measurements been attempted to advance and facilitate modeling efforts. The various shell models are reviewed and discussed. Validation of models of polymer shell agents are given with respect high frequency acoustic microscopy quantification of the shell elastic properties. Recent work on the modeling and design of agents for high frequency subharmonic excitation based on adaptive signal processing and nonlinear time series methodologies is overviewed. Novel analytical method for treating hydrodynamic interactions with respect to unsteady drag and secondary acoustic radiation forces are outlined with respect to drug delivery and sonoporation applications. The direction and magnitude of coupled oscillations are investigated with respect evolution of the amplitude and phase and the transfer entropy.

Contributed Papers

4:10

4pBA8. A two-criterion model for microvascular bioeffects induced *in vivo* by contrast microbubbles exposed to medical ultrasound. Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu) and Douglas L. Miller (Radiology, Univ. of Michigan Health System, Ann Arbor, MI)

The mechanical index (MI) assumes that bubbles of all relevant sizes exist in tissue, yet that assumption is approximated only in studies that include use of a microbubble contrast agent. If the MI is taken to be the key dosimetric parameter, then it should allow science-based safety guidance for contrast-enhanced diagnostic ultrasound. However, theoretical predictions based on the MI typically do not concur with the frequency dependence of experimentally measured thresholds for bioeffects. For example, experimental measurements of thresholds for glomerular capillary hemorrhage in rats infused with contrast microbubbles (Miller *et al*. UMB 2008;34:1678) increase approximately linearly with frequency while the MI assumes a square-root dependence. Here, thresholds for inertial cavitation were computed for linear versions of the acoustic pulses used in that study assuming bubbles containing either air, C_3F_8, or a 1:1 mixture of the two and surrounded by either blood or kidney tissue. While no single threshold criterion was successful, combining results for one criterion that maximized circumferential stress in the capillary wall with another that ensured an inertial collapse, produced thresholds that were consistent with experimental data. This suggests that development of a contrast-specific safety metric may be achieved by further testing and confirmation in different tissues.

4:25

4pBA9. Myocardial cavitation-enabled therapy modeling. Yiying I. Zhu (Biomedical Eng., Univ. of Michigan, 1301 Catherine St., Med Sci I 3218A, Ann Arbor, MI 48109-5667, zhuyiy@umich.edu), Douglas L. Miller, and Oliver D. Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI)

A therapeutic method named Myocardial Cavitation Enabled Therapy aiming at noninvasive cardiac tissue reduction is modeled here. Sparsely distributed microlesions induced by ultrasound cavitation of contrast agents are hypothesized to cause myocardial shrinkage. The objective is to model lesion formation based on the acoustic field and plan treatments accordingly. An ultrasound field simulation was established in Field II, an acoustics toolbox. It simulates the acoustic field of a 1.5 MHz ultrasound burst of five cycles at 4.0 MPa peak rarefactional pressure amplitude (PRPA) by use of a F# 2 single element therapeutic transducer, as used in concomitant animal studies. Medium parameters, including speed of sound, density, absorption, and B/A were set to water path and heart tissue along the acoustic path. Lesions were masked as region exceeding 2 MPa PRPA, the threshold of microlesions occurrence. The lesion volume is 143 μL compared to *in vivo* rodent study of approximately 100 μL. To reach a larger lesion in pigs, sweeping of the ultrasound beam is needed using a phased array. Nonlinear acoustic simulation from k-Wave software will assist transitioning from single element to imaging array apertures in terms of interpreting possibly different lesion formation resulting from nonlinearities.

4:40

4pBA10. Frequency dependence of thresholds for lethal cardiomyocyte injury in myocardial contrast echocardiography. Douglas L. Miller, Xiaofang Lu, and Chunyan Dou (Radiology, Univ Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu)

Contrast enhanced diagnostic ultrasound employs microbubble activation for microvascular imaging; however, the on-screen Mechanical Index is a poor parameter for safety guidance. More research is needed on microvascular bioeffects, particularly their variation with frequency. A GE Vivid 7 with an S3 probe operated at 1.6 MHz, and an S5 probe operated at 2.5 and 3.5 MHz was used for myocardial contrast echocardiography of rats mounted in a water bath. Power settings were varied in 2 dB steps for determination of the thresholds for cardiomyocyte injury. The contrast agent was made to duplicate the properties of the clinical agent Definity. The scans were intermittently triggered each 4 heartbeats from the ECG signal. The cardiomyocyte death was assessed using Evans blue vital staining. Thresholds were defined as the mean of the lowest exposure with a statistically significant cardiomyocyte death and the next lower exposure level. Thresholds were 1.2 MPa, 1.7 MPa, and 2.7 MPa peak rarefactional pressure amplitude (derated for 1 dB/cm/MHz attenuation) for 1.6, 2.5, and 3.5 MHz, respectively. Linear regression showed that the thresholds were essentially proportional to frequency (0.72 f^{1}*.02*, r^2 = 0.97). Theoretical analysis is under development to explain this dependence and develop a contrast-specific safety parameter.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 ORLANDO, 1:00 P.M. TO 2:15 P.M.

Session 4pEA

Engineering Acoustics and Structural Acoustics and Vibration: Layered Media

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg. 1302, Newport, RI 02841

Andrew J. Hull, Cochair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841

Invited Papers

1:00

4pEA1. Acoustic scattering from rib-stiffened finite bilaminar composite cylindrical shells- 3-D solution. Sabih I. Hayek (Eng Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NUWC, Newport, RI)

The acoustic scattering from an insonified finite bilaminar cylindrical shell stiffened by a thin circular rib-stiffener is analyzed. The two cylindrical shell laminates are perfectly bonded having the same lateral dimension but have different radii and material properties. The bilaminar shell is analyzed using the exact theory of three-dimensional elasticity. The thin rib-stiffener has rectangular cross-section and is perfectly bonded to the inside of the inner shell. It is analyzed as a thin elastic ring. The finite shell has shear-diaphragm supports at ends $z=0$ and L and is terminated by two semi-infinite rigid cylindrical baffles. The shell is insonified by an incident plane wave at an oblique incidence angle. The scattered acoustic farfield is evaluated for various incident wave frequencies and stiffener location and dimensions. A uniform steel stiffened shell in water was initially analyzed to study the influence of stiffened-shell geometries on the scattered acoustic farfield. A second shell made up of an outer elastomer shell bonded to an inner stiffened steel shell was also analyzed to study the influence of elastomeric properties on acoustic scattering. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

1:20

4pEA2. Imaging a barely visible impact damage in a laminated composite without contact using air-coupled nonlinear elastic wave spectroscopy. Marcel C. Remillieux (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, mcr1@lanl.gov), Łukasz Pieczonka (Dept. of Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), Pierre-Yves Le Bas, Brian E. Anderson, and TJ Ulrich (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM)

We present the first set of experiments in which air-coupled ultrasonic emission is used for exhaustive and rapid imaging of a Barely Visible Impact Damage (BVID) introduced by an impact test on a laminated composite. Such experiments have been limited thus far by the impedance mismatch of nearly five orders of magnitude between the air and an elastic solid. This limitation was overcome by designing an ultrasonic source capable of generating a focused elastic wave field in the sample with an amplitude sufficiently large to allow the use of nonlinear elastic wave spectroscopy. The images of the damaged area obtained with the proposed apparatus and signal processing

technique reveal the same features, namely a delamination and a crack, as with vibro-thermography and more features than with a C-scan, but remove the need to be in direct contact with the composite, thus reducing the test time by orders of magnitude. [This work was supported by the U.S. Department of Energy through the LANL/LDRD Program and the Fuel Cycle R&D, Used Fuel Disposition (Storage) campaign. The visit of Dr. Pieczonka at LANL was supported by the Foundation for Polish Science (FNP) within the scope of the WELCOME Program—project no. 2010-3/2.]

1:40

4pEA3. A stochastic inverse solution for functionally graded acoustic layered metamaterial validation. Heather Reed (Weidlinger Assoc., Inc., 40 Wall St 18th Fl., New York, NY 10005, heather.reed@wai.com), Jeffrey Cipolla (Weidlinger Assoc., Inc., New York, New York), and Patrick Murray (Weidlinger Assoc., Inc., New York, NY)

Functionally graded acoustic metamaterials (FGAMs) can be designed to have specific waveguide properties. In sonar applications, FGAMs can be tailored to resist incident wave reflection. As these materials do not exist naturally, they must be fabricated by gradually layering manufactured, resulting in a (usually smooth) variation of properties. Validating material properties of FGAMs is difficult with conventional tests, as the distribution of material properties over the layered structure results in a non-unique solution if typical data (e.g., compressive strain) is measured. This talk will demonstrate an approach to characterize the functionally graded material properties by parameterizing how the functionally graded material changes throughout the specimen. Experiments designed to minimize the uncertainty surrounding the FGM model parameters are formulated and evaluated numerically. The FGM model parameters are estimated by Markov chain Monte Carlo so that a probability distribution surrounding each parameter is recovered. Probability distributions enable uncertainty quantification (UQ) surround the validated material parameters. UQ is important as uncertainty surrounding the parameter results directly translates into uncertainty surround the FGM performance. The approach is verified by bootstrap analyses of known FGAM distributions.

Contributed Paper

2:00

4pEA4. Seismic surface wave method for subsurface layered soil exploration. Zhiqu Lu (National Ctr. for Physical Acoust., Univ. of MS, 1 Chucky Mullins, University, MS 38677, zhiqulu@olemiss.edu)

Subsurface soils in the vadose zone are layered structures. Typically, there are three distinctive layers at the depths of a few meters, featuring as a top rigid layer (due to surface crusting and sealing), a middle soft zone (due to soil moisture), and a region with stiffness increasing with depth (due to increased overburden pressure). The properties of subsurface soils in the vadose zone are often altered by natural events (weather and chemical reactions) or cultural activities (compaction). The exploration of the subsurface soils is required for agricultural, environmental, civil engineering, and military applications. In this talk, a seismic surface wave technique is developed, a so-called high frequency multi-channel analysis of surface wave method (HF-MASW). In the method, an electromagnetic shaker is placed on the ground and serves as a seismic source, a vibration sensor (either a laser Doppler vibrometer or an accelerometer) is used to record surface vibrations at multiple locations alone a straight line. Rayleigh waves propagation theory, layered structural modeling, and spectral analysis are applied for processing received signals and conducting inversion. A soil profile in terms of shear wave velocity is determined from the method. In this talk, several applications using MASW method will be reported, including layering delineation with cross-section imaging, weather effects monitoring, fragipan detection, and surface crusting and sealing evaluation.

Session 4pMU

Musical Acoustics: Acoustical Evolution of Musical Instruments

Whitney L. Coyle, Chair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—1:00

Invited Papers

1:05

4pMU1. Acoustical innovations in nineteenth-century violinmaking. Sarah Gilbert (Musicology, Florida State Univ., 122 N. Copeland St., Tallahassee, FL 32304, smg11b@my.fsu.edu)

Tensions between innovation and tradition in violinmaking have impeded the acceptance of most attempts to improve or alter the structure of the instrument. The nineteenth century, however, saw a proliferation of innovative violins as luthiers responded to musical developments and changing social and economic environments during the Industrial Revolution. As nineteenth-century composers called for greater range and diversity in timbre, chromaticism, dynamics, range, and key, instruments were developed to accommodate these demands. But perhaps more important than the purely musical considerations was the interdisciplinary collaboration between musicians and scientists in the pursuit of acoustic perfection. Many luthiers, such as François Chanot and Jean-Baptiste Vuillaume, viewed themselves as scientists and engineers, experimenting with acoustic properties and new materials in order to improve upon the existing form of the violin. In a reciprocal relationship, acousticians recognized musical instruments as rich sources for the study of acoustic principles, and luthiers consulted with acousticians and engineers about the technical construction of experimental forms. In this paper, I examine the technical construction of several of these instruments for insight into their novel construction techniques and acoustical properties, relating this experimental trend to the alliance of the sciences and arts during the Industrial Revolution.

1:25

4pMU2. Air resonance power efficiency evolved in the violin and its ancestors. Hadi T. Nia, Ankita D. Jain, Yuming Liu, Mohammad-Reza Alam (Dept. of Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., 5-212, Cambridge, MA 02139), Roman Barnas (Violin Making, North Bennet St. School, Boston, MA), and Nicholas C. Makris (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA, makris@mit.edu)

Acoustic radiation from a violin can be explained by clear physics at its lowest frequency resonance (air resonance), which also helps to explain key aspects of violin design evolution. It is found that inefficient, inactive void area was decimated and air-resonance power doubled as sound-hole geometry of the violin's ancestors slowly evolved over eight centuries from simple circles of Medieval 10th century fitheles to complex f-holes of the late Renaissance and Baroque period. F-hole length then increased across centuries in the renowned Amati, Stradivari, and Guarneri workshops, favoring correspondingly higher air-resonance power, by processes consistent with random craftsmanship mutations and subsequent selection.

1:45

4pMU3. Acoustical evolution of the viola da gamba. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk), J. Patricia Campbell (Edinburgh College of Art, Univ. of Edinburgh, Edinburgh, United Kingdom), and Jim Woodhouse (Dept. of Eng., Univ. of Cambridge, Cambridge, United Kingdom)

The viola da gamba (or viol) emerged as a distinctive type of bowed string instrument during the fifteenth century. Its characteristics include a fingerboard with gut frets and between five and seven strings tuned in fourths with a central major third. By the end of the fifteenth century, the instrument was being constructed in a range of sizes. To investigate the acoustical significance of design changes during the evolution of the viol family in the sixteenth and seventeenth centuries admittance measurements at the bridge have been made on a wide variety of reproductions of viols and other early bowed-string instruments. A comparison of the results with existing data on modern violin-family instruments, and other stringed instruments such as guitars, reveals that the acoustic behavior of viols is more disparate than that of violins or guitars. Models that have been measured include examples with and without a soundpost, and a correspondingly wide range of frequency responses is seen. The influence of the viol bridge and the significance of the adoption of overwound bass strings are also discussed.

2:05

4pMU4. Early history of the European free reed instruments. James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, jcotting@coe.edu)

Free reed instruments have been known in Asia for thousands of years, but the Western free reed instruments such as the harmonica, accordion, and reed organ were only invented and developed during the last two centuries. In 1780, Kratzenstein published a paper in St. Petersburg describing a speaking machine, which produced vowel sounds using free reeds with resonators of various shapes. This event marks a convenient, if arbitrary, starting point for the history of the free reed musical instruments of European origin. These instruments developed rapidly, and by 1850, the accordion, concertina, harmonica, reed organ, and harmonium all had been invented and developed into more or less final form. This paper presents some episodes in the development of these instruments. Also addressed is the question of the influence of the Asian free reed mouth organs on the origin and development of the European free reeds.

2:25

4pMU5. Evolution of the piano. Nicholas Giordano (Phys., Auburn Univ., 246 Sci. Ctr. ClassRm. Bldg., Auburn, AL 36849-5319, njg0003@auburn.edu)

The first pianos were built around 1700 and unlike the case with many other instruments, we know precisely what the first pianos were like since several of them still exist. These early pianos were patterned after Italian harpsichords of the time, with the plucking mechanism of the harpsichord replaced by an "action" mechanism that enabled hammers to strike the strings. The first pianos had a compass of four octaves with brass and iron strings, and a frame composed of wood. The design of those instruments along with the various stages of their evolution into the modern piano with its 88 notes, steel strings and a cast iron plate will be described. The manner in which this evolution was driven by the needs of composers and musicians along with advances in technology will also be discussed.

2:45–3:00 Break

3:00

4pMU6. The acoustic complexity of keyboard percussion instruments. Part I: (What we know and what we wish we knew) Leigh Howard Stevens. Leigh H. Stevens (21 South Riverside Dr., Neptune, NJ 07753, Drsplatbar@Gmail.com)

Part I Introduction to tuned bars and resonators Keyboard Percussion Instruments include innumerable variations of marimbas, xylophones, vibraphones, and glockenspiels. The tone bars, made of wood, metal, or other materials, are often incapable of producing sufficient volume without the amplifying assistance of a resonating tube or chamber. Unfortunately, these two vibrating systems (bar and resonator) react in opposite directions to changes in temperature, decoupling from their ideal relationship with any rise or fall in temperature. Also, unlike string, brass, or woodwind instruments, where the overtones are produced "naturally" and "are what they are," the overtones of the bars of keyboard percussion instruments can be manipulated and controlled in the design and manufacturing process. For example, while both instruments have tone bars made of rosewood, the xylophone has the second partial tuned to an octave and a perfect fifth above the fundamental, while a marimba has the second partial tuned to a frequency two octaves above the fundamental. While we can identify and lower the frequency of many of the prominent partials contained in tone bars, their interdependence within the bar, and their coupling with a resonator provides a level of complexity for the designer and manufacturer, perhaps unrivaled in musical instrument manufacturing.

3:20

4pMU7. The acoustic complexity of keyboard percussion instruments. Part II: What we know and what we wish we knew Leigh Howard Stevens. Leigh H. Stevens (21 South Riverside Dr., Neptune, NJ 07753, Drsplatbar@Gmail.com)

Part II Resonating chamber shapes, their Q factor, their resultant musical character Resonating Chambers come in many shapes and sizes and largely determine the power, focus, and sustain of whatever the associated tone bar is coupled with it. The most basic box resonators are found on elementary education musical instruments such as Carl Orff tuned percussion instruments: demonstration of various tone bars resonating over box, rectangular, square and round tube resonators; demonstration of high and low Q systems responding to the same tone bar; demonstration of well-tuned, miss-tuned, and deliberately de-tuned partials on various tone bars; and questions from (and challenges to!) the audience.

3:40

4pMU8. A short acoustical history of flutes from the Paleolithic period to nowadays. Benoit FABRE and Camille Vauthrin (Sorbonne Universités, UPMC Univ Paris 06, UMR 7190, Institut Jean le Rond d'Alembert, 75015, Paris, France, CNRS, UMR 7190, Institut Jean le Rond d'Alembert, 11, rue de Lourmel, 75015 Paris, France, UPMC-LAM d'Alembert, LAM d'Alembert, Sorbonne Universités, UPMC Univ Paris 06, UMR 7190, Institut Jean le Rond d'Alembert, 75015, Paris, France, vauthrin@lam.jussieu.fr)

Flutes appear at different time periods and places over the world. The instrument making adapts to the cultural and musical context. Studying the instruments from the acoustical point of view allows to understand some evolutions in instrument making, focusing on different aspects such as the geometry of the bore of the instrument and its consequences on the passive resonances, the playing conditions and aeroacoustical consequences, the sound characteristics such as intonation, spectral content, and sound intensity. The talk will focus on a few instruments and periods in order to illustrate some evolutions and relations between these different aspects.

4:00

4pMU9. The clarinet: Past, present, and future. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu)

The modern clarinet is the result of several hundreds of years of research and craftsmanship. This paper will discuss where the clarinet has been acoustically by studying some input impedance spectrum characteristics of historical instruments (the Baroque Chalumeau, 13-key, etc.). The paper will then present where we are at present—focusing on the same measurement information for a full Boehm system (multiple models of Bb clarinets) and a comparison of the modern French vs. German system clarinet. Finally, a few words will be given on the possible future of the instrument—where we are going, what are we doing to continue improving the instrument's quality and playability.

4:20

4pMU10. Acoustical and metalworking techniques study of cornua, roman brass instruments, and their sound reproduction by physical modeling sound synthesis. René E. Caussé (Ircam - UMR STMS CNRS - UPMC, 1 Pl. Igor Stravinsky, Paris 75004, France, Rene.Causse@ircam.fr), Benoît Mille (Ctr. de Recherche et de Restauration des Musées de France (C2RMF), Paris, France), and Margaux Tansu (Ecole nationale supérieure de chimie de Rennes (ENSCR), Rennes, France)

Under the musical instrument analysis project of ancient societies, combining musical archaeologists and research laboratories on materials and acoustic, a first technological study was conducted in the storage rooms of the Naples Archaeological museum. The five instruments were incomplete and consist of several pieces assembled on a plexiglass tube that does not allow the direct measurement of the bore. Two mouthpieces were associated with these instruments. Morphometric study was conducted and detached fragments of the Cornua were collected. The corresponding samples were then analyzed under a Bright Field Optical Microscope (BFOM) and a Scanning Electron Microscope (SEM) in order to determine the metalworking techniques. To determine the elemental composition of the Cornua (alloying and trace elements), analyses by means of Particle Induced X-ray Emission (PIXE) using the Ion beam facility AGLAE at C2RMF. From the morphometric data, the bore of the different cornua was reconstruct and their resonant frequencies calculated. Several possibilities of reconstituting the bore were tested and compared. Finally, the resonances obtained from the reconstructions of these instruments of the past will be compared with those of present instruments like French horn and trombone. The various reconstructions will also be compared from sound examples obtained through physical modeling synthesis.

Contributed Paper

4:40

4pMU11. The Tintignac carnyx: An acoustical study of an early brass-wind instrument. Michael Newton (Acoust. and Audio Group, Univ. of Edinburgh, James Clerk Maxwell Bldg., Edinburgh EH9 3FD, United Kingdom, michael.newton@ed.ac.uk), John Kenny (Royal Conservatoire of Scotland, Glasgow, United Kingdom), John Chick, Amaya Lopez-Carromero, D. Murray Campbell (Acoust. and Audio Group, Univ. of Edinburgh, Edinburgh, United Kingdom), and Joel Gilbert (Laboratoire d'Acoustique de l'Université du Maine, Le Mans, France)

The carnyx was a metal wind instrument used by Celtic peoples around two thousand years ago. It was approximately two meters long with a bell in the shape of an animal head. In 2004, an excavation at Tintignac in the Corrèze district of France uncovered a horde of bronze instruments, including parts of several carnyxes. It proved possible to assemble an almost complete carnyx from these parts, and in 2011, a copy of this carnyx was made in brass by Jean Boisserie. The acoustical behavior of the brass copy was studied by Joel Gilbert and colleagues at the Université du Maine in Le Mans; this work led to the proposal that the musical performance of the instrument would be improved if an additional section of tubing was included in the reconstruction. Later, Boisserie made a second copy in bronze, together with several optional extension pieces. The musical performing possibilities of the bronze copy have been studied by the musician John Kenny, and its acoustical behavior has been studied at the University of Edinburgh. The results of these studies, including measurements of sound radiation by the large bronze ears attached to the head, are presented and discussed.

Session 4pNS

Noise, Physical Acoustics, and Animal Bioacoustics: Wind Turbine Noise

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

Kenneth Kaliski, Cochair
RSG Inc, 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics, Wellesley, MA 02482

Chair's Introduction—1:15

Invited Papers

1:20

4pNS1. A proposed test of some people's ability to sense wind turbines without hearing or seeing them. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

Around the world, there are people that allege they sense times when significant power is being generated by a turbine from when little to no power is being generated. In contrast, the American Wind Energy Association (AWEA) and its Canadian counterpart (CanWEA) maintain that this is not possible, without these people hearing the turbines or seeing them move. A simple test can quickly clarify these opposing views. Notionally, people who live in residences situated beyond the threshold of hearing (indoors) would be tested in the area that they can sense the wind turbines. They would be told to tell the proctor when they sensed the turbine at power. They would be told the turbine might or might not be turned on in the next X hours. The area of the house they were in would have to be free of any visible or audible clues. No communication would be permitted between the subject and the outside world. The proctor also would not know when or if the turbine would come on. These and other controls like these would make for a quick, simple, unbiased test of some people's ability to sense wind turbines without hearing or seeing them.

1:40

4pNS2. Advanced signal processing techniques and optimized measurements to extract wind turbine signatures in nearby homes. Andy Metelka (Sound and Vib. Solutions Canada, Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Previous measurements in homes near wind turbines indicate higher pressure levels below 10 Hz than audible pressure levels measured at the same time and location (ASA, vol. 20, 2013 Dooley & Metelka). Blade pass harmonic pressures were proven to be higher relative to basement ground-borne vibration and seismic vibration in floors throughout the home are reexamined simultaneous to broadband pressure. Although the pressure vs. frequency distribution appear to be the same at different locations in one room, they differ when compared to other rooms due to room dimensions. Further multichannel signal processing using a variety of different sensors at various locations inside a home identify areas in a home that are least affected by wind turbine low frequency discrete harmonics as well as acoustic metrics. Wind direction, wind speed, and other factors are precisely measured simultaneously. Sound level metrics inside homes are also compared to narrowband FFT data in an attempt to quantify annoyance and derive alternate metrics.

Contributed Papers

2:00

4pNS3. Radial mode analysis of broadband noise in flow ducts using azimuthal sensor array. Kunbo Xu (School of Power and Energy, Northwestern PolyTech. Univ., No. 127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

For the evaluation and improvement of noise reduction notions and the verification of broadband sound power measurement in flow ducts, it is interesting to obtain the detailed information of the in-duct acoustic mode spectrum and subsequent broadband noise sources separation. A new broadband noise mode analysis method, which uses full wall-mounted sensor array, was experimentally applied on broadband sound fields at three operation conditions, which were generated by means of high loaded single-stage axial flow fan test rig. Two axial sensor arrays were mounted wall-flush upstream of the fan. Measurements were made at operating conditions from 60% to 100% rotor design speed. On the whole, the new method behaves robustly. It delivers physically meaningful broadband mode amplitudes. Mode coherence functions are calculated between all pairs of propagating modes. This feature enables the detailed comparison of different sound fields with characteristically coupled mode pairs. For tonal noise, mode coherence results show that modes are correlated with the source and with each other, especially at blade passing frequencies. The experimental outcome proves the usefulness of the analysis technique for interpreting and understanding broadband sound propagation in turbomachinery flow ducts.

2:15–2:30 Break

2:30–5:00 Panel Discussion

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — ST. JOHNS, 1:35 P.M. TO 4:25 P.M.

Session 4pPA

Physical Acoustics and Noise: Launch Vehicle Acoustics II: Analysis and Modeling of Noise from Supersonic Jets

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Richard L. McKinley, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—1:35

Invited Papers

4p THU. PM

1:40

4pPA1. Acoustic field of a shielded rectangular supersonic jet. Ephraim Gutmark, Pablo Mora, and Florian Baier (Aerosp. Eng., Univ. of Cincinnati, Rhodes Hall 799, Cincinnati, OH 45221-0070, ephraim.gutmark@uc.edu)

The impact of a flat surface installed parallel to a supersonic rectangular jet of AR = 2 on the far field noise is studied. Far-field cold jet results at design, over-expanded and under-expanded conditions are compared between the free-field jet, the nozzle with the surface matching with the exit lip ($h = 0D_e$), and the surface at different stand-off positions away from the jet axis, h = 1, 2, 3 D_e. Results are shown for all jet azimuthal angles. When the surface is installed at $h = 0D_e$, broadband shock-associated noise intensity was decreased and its peak frequency shifted. At NPR = 2.5, shock noise appears to be entirely mitigated. Also, a low frequency noise component is observe below St = 0.15, and assumed to be related to the trailing edge/jet plume interaction. At NPR = 3.0, strong screech tones are mitigated with the surface installed at $h = 0D_e$. In the shielded region, noise levels are significantly lower for all plate positions. Mixing noise and Mach wave radiation are affected by the plat stand-off location at the $\phi = 90^o$. Heated jet measurements up TR = 3.0, near-field and high-speed shadowgraph visualization are performed and reported.

2:00

4pPA2. Modeling of fighter jet cockpit noise. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Hilary L. Gallagher, and Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson Air Force Base, OH)

Military fighter jet cockpit noise may pose a hearing loss risk to pilots and can disrupt communications. The current study is an attempt to model the noise levels in the cockpit of F-35 aircraft as a function of aircraft operation parameters. Levels are based on a complex interplay of parameters such as airspeed, altitude, dynamic pressure, and the flow rate of the in-cockpit environmental control system. Full population of an operational envelope from point-by-point measurements would require an excessive number of flights. In the current work, in-cockpit noise levels were measured during sampled flight conditions. Correlations between levels and flight parameters were investigated. A preliminary parameter-based model is presented, and the ability to predict levels within the operational flight envelope is addressed. [Work supported by USAFRL through ORISE.]

2:20

4pPA3. Development of hearing protection device calculators for the F-35A and F-35B tactical aircraft. Anthony R. Pilon (Adv. Dev. Programs, Lockheed Martin Aeronautics Co., 1011 Lockheed Way, Palmdale, CA 93599-1100, tony.pilon@lmco.com)

Simple spreadsheet-based near field noise prediction tools for the F-35A (conventional takeoff and landing) and F-35B (short takeoff and vertical landing) aircraft have been developed recently. Near field noise predictions are based on empirical data collected during static engine runs conducted at Edwards AFB, and during hover/vertical landing operations at MCAS Yuma in September, 2013. The tools are programmed to calculate near field one-third octave band sound pressure levels as a function of engine power setting and listener location. The calculated near field levels can be used to determine the *Total Daily Exposure (TDE)* of personnel required to work in the vicinity of powered aircraft. Alternatively, the calculators can be used to determine the amount of hearing protection required so that the personnel TDEs remain below 1.0. The calculators are adjustable so that they can be employed for differing domestic and international noise exposure regulations. [*Work supported by the Air Force Research Laboratory and Ball Aerospace*.]

2:40

4pPA4. Jet noise of high-performance aircraft at afterburner. Christopher Tam (Mathematics, Florida State Univ., 1017 Academic Way, Tallahassee, FL 323064510, tam@math.fsu.edu)

The jet noise from a high-performance aircraft at afterburner is investigated. The main objective is to determine whether the dominant noise components are the same or similar to those of a hot supersonic laboratory jet. For this purpose, measured noise data from F-22A Raptors are analyzed. It is found, based on both spectral and directivity data, that there is a new dominant noise component in addition to the usual turbulent mixing noise. The characteristic features of the new noise component are identified. Measured data indicates that the new noise component is observed only when the rate of fuel burn of the engine is increased significantly above that of the intermediate power setting. This suggests that the new noise component is combustion related. The possibility that it is indirect combustion noise generated by the passage of hot spots from the afterburner through the nozzle of the jet is investigated.

3:00–3:20 Break

3:20

4pPA5. Source characterization of full-scale jet noise using vector intensity. Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., 688 North 500 East, Provo, UT 84606, titorep@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Wright-Patterson AFB, Air Force Res. Lab., Dayton, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Vector acoustic intensity has the benefit over pressure measurements in that both the direction and magnitude of energy flow are represented. However, this important quantity has seen little application previously in aeroacoustics. In the present work, an intensity probe captured the radiated field to the sideline and aft of a tethered, full-scale military jet aircraft as one engine was operated at multiple engine conditions. Intensity data from each probe location provide a frequency-dependent map of the sound flow near the aircraft. The vector acoustic intensity is estimated using a recently developed processing technique that extends the traditional upper-frequency limit on estimation accuracy. The intensity vectors are traced back to the jet centerline as a method of approximating the extent and location of the source region as a function of frequency. This method is validated by numerical simulation. As expected for jet mixing noise sources, the resulting source region estimates contract and move upstream with increasing frequency. In addition, the source region at afterburner compared to full-throttle is located about 1 m farther downstream at frequencies above about 300 Hz, and the intensity tends to point more to the sideline by up to 10°. [Work supported by ONR.]

Contributed Papers

3:40

4pPA6. Analytical intensity calculated from a wavepacket model and comparison to intensity measurements near a high-performance military aircraft. Eric B. Whiting, Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, ericbenwhiting@gmail.com), Alan T. Wall (Air Force Res. Lab., Dayton, OH), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

To create an equivalent source description of jet noise, a wavepacket model is used to determine the acoustic vector intensity away from the source. This paper represents an initial investigation into using the measured vector acoustic intensity to define wavepacket parameters. The complex pressure of a line source is defined according to an analytical hyperbolic tangent wavepacket model and Rayleigh integration is used to find the pressure, particle velocity, and time-averaged intensity at observer locations. The parameters that define the shape of the wavepacket are initially based on prior pressure level-based optimization carried out for ground-based microphones. The resulting calculated acoustic intensity vectors are compared to vector intensity measurements previously taken near a tethered high-performance military aircraft at military and afterburner engine conditions. The wavepacket parameters are then varied to provide an optimal fit using the intensity, rather than pressure level, as the cost function, and the differences between the two models are described. [Work supported by the Office of Naval Research.]

3:55

4pPA7. Comparison of holography, beamforming, and intensity-based inverse measurements on full-scale jet noise sources. Part I: Numerical investigations. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Blaine M. Harker, Trevor A. Stout, Tracianne B. Neilsen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The reduction of high-performance fighter aircraft noise, which is of particular concern due to its potential negative impact on communities near air bases and damage to military personnel hearing, is facilitated by improved understanding of the jet noise sources. To this end, near-field acoustical holography, beamforming, and acoustic vector intensity methods have been investigated for the source imaging of full-scale jets. In the first of a two-part investigation, a numerical experiment is performed to test the accuracy of each of these methods, using a simulated measurement array similar to a physical array from an actual full-scale measurement. Extended, partially correlated sources are generated over a range of frequencies using a wavepacket ansatz, and array measurements are simulated in the geometric near field. Source reconstructions are obtained using each of the methods. These equivalent sources are then propagated to the mid and far field. The various reconstructions are compared to simulated benchmarks in order to highlight the advantages and disadvantages of each method, and to evaluate the frequency ranges over which they provide accurate results at the source and in the mid and far fields. [Work supported by USAFRL through ORISE.]

4:10

4pPA8. Comparison of holography, beamforming, and intensity-based inverse measurements on full-scale jet noise sources. Part II: Experimental results. Blaine M. Harker (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Trevor A. Stout, Tracianne B. Neilsen, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Targeted reduction of high-amplitude jet noise is facilitated by accurate source imaging and sound field prediction models. To provide guidance for future jet noise reduction efforts, near-field acoustical holography, beamforming, and acoustic vector intensity-based inverse methods (AVIBIM) have been implemented in efforts to measure and predict full-scale high-performance aircraft noise sources. Patch-and-scan array measurements were taken in the near-field of a tethered tactical aircraft. From these data, equivalent source reconstructions are calculated using each method at multiple frequencies, and the source results are then propagated outward and compared with benchmark locations in the mid and far fields. The frequency ranges where each method's reconstructions predict the measured sound pressure levels are identified and guidelines are provided for the design of measurements to optimize source reconstructions. These results are compared to those obtained in the numerical study (Part I) of this two-part investigation. [Work supported by USAFRL through ORISE.]

Session 4pPP

Psychological and Physiological Acoustics: Pitch, Loudness, and Other Perceptual Phenomena

William M. Hartmann, Chair
Physics and Astronomy, Michigan State University, Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824

Contributed Papers

1:30

4pPP1. The perception of vocal-tract length in cochlear implant users: Can it be improved? Etienne Gaudrain (Cognition Auditive et Psychoacoustique, Ctr. de Recherche en NeuroSci. de Lyon, CNRS UMR 5292, Université Lyon 1, UMCG, KNO, Huispostcode BB20, PO box 30.001, Groningen 9700 RB, Netherlands, etienne.gaudrain@cnrs.fr), Nawal El Boghdady, and Deniz Başkent (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

When trying to understand a speaker among competing speakers, normal-hearing listeners take advantage of differences in the voices of the talkers. However, cochlear implant (CI) users do not seem to benefit from such voice differences. Roy D. Patterson's group in Cambridge has applied psychophysical methods to voice perception in order to isolate the acoustic cues that allow listeners to identify and segregate voices [e.g. Smith, Patterson, Turner, Kawahara and Irino, 2005, J. Acoust. Soc. Am., vol. **117**, pp. 305–318]. In our group, we have extended these methods to CI listeners to fully characterize the limitations CI users experience in utilizing voice characteristics. More specifically, we have shown that while voice pitch (F0) is sufficiently salient through the implant to allow gender categorization, the perception of vocal-tract length (VTL) is severely degraded, leading to erroneous gender perception. With the help of acoustic simulations mimicking the CI processing, the potential factors explaining this lack of sensitivity for VTL were explored. Poor spectral resolution, spectral quantization, and spectral distortion were all found to affect VTL just-noticeable-differences. These results have yielded a number of potential solutions to improve VTL representation — and hence concurrent speech perception — in the implant.

1:45

4pPP2. Probing monaural edge pitch. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu), Xiaochen Li (Dept. of Phys., Xian Jiaotong Univ., Xian, China), and Peter Cariani (Biomedical Eng., Boston Univ., Boston, MA)

Noise with a sharp spectral edge, occurring at an edge frequency f_e, leads to a tonal sensation with a timbre like that of a sine tone and a pitch near the edge frequency. For lowpass noise the pitch is shifted below f_e and for high pass noise the pitch is shifted above. Typical shifts are less than 10% of the edge frequency. To probe this effect, pitch matching experiments were done using 32 different lowpass and highpass noises with edges at 500 and 600 Hz and with durations of about 50 ms. Pitch matches were made in an indefinitely repeating cycle of noises and half-second matching tones. After each match, listeners rated the difficulty of the match. The ratings were ultimately reduced to a five-point scale. The experimental data were compared with the predictions of a model based on the peaks of an all-order population interval distribution as determined by the Zilany-Bruce-Carney model of the auditory periphery. Matching frequencies were predicted from the periodicities of the best fitting subharmonic template and matching difficulty was predicted by the relative goodness of fit. Correlations between experiment and prediction were often high, approaching 0.9. [Work supported by the USAFOSR].

2:00

4pPP3. Acoustic and auditory sketches: Recognition of severely simplified natural sounds by human listeners. Vincent Isnard (Département Action et cognition en situation opérationnelle, Institut de recherche biomédicale des armées, 1 Pl. Igor-Stravinsky, Paris 75004, France, vincent.isnard@ircam.fr), Marine Taffou, Isabelle Viaud-Delmon (Espaces acoustiques et cognitifs, Institut de recherche et coordination acoustique/musique, Paris, France), and Clara Suied (Département Action et cognition en situation opérationnelle, Institut de recherche biomédicale des armées, Brétigny-sur-Orge, France)

Sounds in our environment like voices, animal calls, or musical instruments can be recognized on the basis of timbre alone. The objective of this study was to unravel the features underlying this robust sound recognition. We investigated how severely sounds can be simplified while still being recognizable. A large number of natural sounds were simplified into "auditory or acoustic sketches:" energy peaks were selected on the auditory or acoustic spectrogram, at three levels of simplification (high, medium, and low). The remaining non-sparse information was removed. Listeners were asked to classify the simplified sounds into their four original categories: instruments, birds, vehicles, or voices. We adapted a recent signal detection model to dissociate the perceptual sensitivity (d' scores) from the bias, for each category (4-AFC task). Overall, severely simplified sounds could still be recognized above chance by the listeners. Auditory distances, based on spectro-temporal excitation patterns (STEPs), were then computed. Participants' performances on the 4-AFC task were well correlated with the auditory distances between the four categories. Altogether, our results suggest that sound recognition is a very robust perceptual process, and that basic spectral and spectro-temporal differences between sounds, captured by the auditory distances, can account for this robust recognition.

2:15

4pPP4. Loudness effect on pairwise comparisons and sorting. Patrick Susini, Olivier Houix (R&D, IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, susini@ircam.fr), and Guillaume Saint Pierre (IFSTTAR, Paris, France)

Effect of loudness on the perceptual structure underlying a corpus of sounds is investigated by two experimental methods: pairwise comparisons and sorting. Both methods are applied to a corpus of recordings sounds presented with their ecological, non-normalized loudness, and to the same corpus equalized in loudness. Two types of perceptual structures (multidimensional scaling and hierarchical cluster analysis) are derived. Domination of one auditory attribute—loudness—on less salient ones is discussed according to the two types of perceptual structures. In the non-loudness-normalized corpus, loudness was the main factor explaining participant's judgments for both tasks. In particular, representations derived from sorting data almost solely reflected sound pressure level difference between sounds. On the opposite, loudness normalization allowed the emergence of several predicted auditory attributes that characterize the tested sounds. On that second corpus however, sorting data were found less appropriate than pairwise data to provide interpretable continuous dimensions.

2:30

4pPP5. Neuromagnetic correlates of the vocal characteristics of vowels in auditory cortex. Roy D. Patterson (Physiol., Development and Neurosci., Univ. of Cambridge, Downing Site, Cambridge CB2 3EG, United Kingdom, rdp1@cam.ac.uk), Martin Andermann (Section of Biomagnetism, Dept. of Neurology, Univ. of Heidelberg, Heidelberg, Germany), Stefan Uppenkamp (Medizinische Physik, Univ. of Oldenburg, Oldenburg, Germany), and André Rupp (Section of Biomagnetism, Dept. of Neurology, Univ. of Heidelberg, Heidelberg, Germany)

As a child grows up, the formants of their vowels move down in frequency and the spacing of the harmonics, which determines voice pitch, decreases. Perceptually, these variables determine who we hear speaking (child, woman, or man). In a logarithmic-frequency spectrum, the envelope with its formant peaks moves toward the origin as a unit, without changing shape, as a child grows up. Similarly, the harmonics, which constitute the fine structure of the spectrum, move toward the origin as a unit without changing shape, but at a different rate. This paper describes neuromagnetic studies which show that the generator associated with voice pitch is in Heschl's gyrus just lateral to primary auditory cortex, while the generator associated spectral envelope position is in planum temporale, some distance behind the pitch activity generator. The posterior generator is close to the location of the large N1m that typically accompanies the onset of acoustic energy of any sort. The pitch processing component of the N1m was isolated from the energy-onset component by presenting sequences of vowels with minimal inter-vowel intervals; the pitch component appears isolated in the responses to the second and succeeding vowels.

2:45–3:00 Break

3:00

4pPP6. Incidental absolute pitch learning in an interactive multi-modal environment. Shannon L. Heald, Stephen C. Van Hedger, and Howard C. Nusbaum (Psych., Univ. of Chicago, 5848 S. University Ave., c/o Shannon Heald (B406), Chicago, IL 60637, smheald@gmail.com)

Absolute pitch (AP), the ability to label an isolated note without the aid of a reference note, is to some degree trainable in an adult population. Most training studies ask participants to make overt categorization judgments and provide explicit feedback. Here, we examined the incidental learning of four isolated notes in an interactive multi-modal environment. Participants played a video game, whereby four characters, requiring different actions, were marked by specific note classes. Half of the participants experienced an invariant mapping, such that the same perfectly in-tune note always marked each character. The other half experienced a variant mapping, where a distribution of notes that varied in intonation marked each character (the mode of which was an in-tune note). While participants could anticipate a character via its associated note class, such anticipation was not required or encouraged. In a follow-up task, participants matched the characters to their associated notes. Results showed significant rote learning for the character-note pairings with above-chance generalization to novel timbres and octaves. The variant condition mitigated the performance gap found between those with and without video game experience in the invariant condition. Results will be discussed in terms of the mechanisms involved in implicitly acquiring AP categories.

3:15

4pPP7. Some factors influencing loudness asymmetries between rising and falling-intensity stimuli. Emmanuel Ponsot (IRCAM, Paris, France), Sabine Meunier (LMA-CNRS, Marseille, France), and Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, susini@ircam.fr)

Previous research demonstrated that the loudness asymmetry between 1-kHz rising and falling-intensity tones is a robust phenomenon, whose origins still remain unclear. In the present study, this phenomenon was further examined as a function of two stimuli characteristics: the spectral content and the intensity-region. In a first experiment, the global loudness of rising and falling-intensity sounds with various spectral contents (pure tones from 250 Hz to 8 kHz and broadband noises) presented in different intensity-regions (from [50–65 dB SPL] to [70–85 dB SPL]) was assessed in an absolute magnitude estimation task. Significant asymmetries were found for tones at all frequencies, but not for broadband noises. In addition, a significant interaction between the stimulus direction and the intensity-region was observed for both tones and noises. This interaction was further examined in a second experiment using an adaptive loudness-matching procedure. Although greater asymmetries were again observed for tones, significant asymmetries were found for noises as well. Furthermore, the size of the asymmetries was significantly decreased with the intensity-region when the pairs were composed of rising followed by falling stimuli. These results are discussed in the light of recent physiological and neuroscience studies conducted with this type of stimuli.

3:30

4pPP8. The perception of vocal characteristics in normal-hearing and cochlear-implanted children. Deniz Baskent (UMCG Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, PO Box 30.001, Groningen 9700RB, Netherlands, d.baskent@umcg.nl), Jacqueline Libert (Res. School of Behavioral and Cognit. Neurosci., Univ. of Groningen, Groningen, Netherlands), Deborah Vickers (Ear Inst., Univ. College London, London, United Kingdom), and Etienne Gaudrain (Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Voice characteristics provide important cues for many speech-related tasks, such as gender categorization, vocal emotion perception, and understanding speech in the presence of competing voices; tasks known to be difficult for cochlear implant (CI) users. While most related research has focused on voice fundamental frequency (F0), the work from Smith *et al.* [J. Acoust. Soc. Am., vol. **117**, pp. 305–318, 2005] has shown the importance of another voice characteristic: the vocal tract length (VTL; related to speaker size). Recent work has indicated that while CI users have difficulty in utilizing F0, they are even more limited at VTL perception. This study aimed to further understand factors potentially contributing to this limitation, by exploring F0 and VTL perception in NH children and children with CIs. We hypothesized that children may learn to utilize F0 before VTL cues because of exposure to exaggerated F0 cues in infant-directed-speech, and acquisition of VTL cues may be later, due to the necessity for exposure to multiple talkers; implying a hierarchy in processing the two vocal cues. We further hypothesized that children with CIs may be better at processing voice cues than adult-implanted CI users, suggesting that these perceptual cues could be learned.

3:45

4pPP9. Sound source localization for filtered noises when listeners rotate. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Yost and Zhong (2015, JASA, 137, 2200) recently showed that sound source localization when listeners rotate depends on both spatial cues and information about the world-centric location of the listener. In these experiments, the sound was a broad-band (0.125—15 kHz), 200-ms noise burst. This stimulus does not allow for an estimation of the role of interaural time (ITD) and/or level (ILD) differences in sound source localization when listeners rotate. The present experiment used the following stimuli: low-pass noise (0.125–0.5 kHz) implicating ITD cues, high-pass noise (2–8 kHz) implicating ILD cues, broad-band noise (0.125–8 kHz) implicating both ITD and ILD cues, and mid-frequency noise (1–4 kHz) for which neither an ITD nor an ILD cue provides good information about sound source location. Listeners rotated at constant velocity (45o/sec) in the azimuth plane, while sounds changed position around a 24-loudspeaker array in the same azimuth plane. Listeners responded in an eyes-open condition in which they would receive information about their world-centric location and in an eyes-closed condition in which they would not. The task was to decide if two successive noise bursts were presented from the same or different loudspeakers as listeners rotated. [Research funded by an AFOSR grant.]

4:00

4pPP10. Verification of an automated headphone-based test of spatial release from masking. Frederick J. Gallun (Otolaryngology/Head&Neck Surgery, Oregon Health and Sci. Univ., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Sean D. Kampel (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Kasey M. Jakien, Nirmal K. Srinivasan (Otolaryngology/Head&Neck Surgery, Oregon Health and Sci. Univ., Portland, OR), Meghan M. Stansell, and Samuel Y. Gordon (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR)

Currently, there are many different laboratory-based tests of spatial release from masking (SRM) that use speech materials; however, there is still disagreement as to the impact of age and hearing loss on SRM. The time is ripe, then, for taking these tests out of the laboratory and testing larger numbers of listeners varying in age and hearing ability in order to provide the statistical power needed to answer the questions currently being asked. Unfortunately, most of the tests that have been developed are either open set, and thus require a tester to administer them, or require complex soundfield speaker arrays. Our laboratory has recently developed and verified an automated headphone-based test that can be presented in only five to ten minutes and that provides results that are predictive of results obtained in an anechoic chamber. The data associated with the verification of this test procedure will be presented.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — DAYTONA, 2:45 P.M. TO 4:35 P.M.

Session 4pSA

Structural Acoustics and Vibration: Novel Treatments in Vibration Damping

Kenneth Cunefare, Chair
Georgia Tech, Mechanical Engineering, Atlanta, GA 30332-0405

Invited Papers

2:45

4pSA1. Vibration damping and isolation using negative stiffness structures. Michael R. Haberman, Carolyn C. Seepersad, and Preston S. Wilson (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

This work presents a novel class of engineered structures with significant promise to improve vibration damping treatments and isolation systems: negative stiffness (NS) elements. A mechanical system displaying negative stiffness is characterized by a loading state that requires a decreasing force level to increase the deformation of the system. Systems displaying NS will possess regions of negative curvature in their strain energy response as a function of deformation; hence, they are unstable when unconstrained. Analytical and experimental results will be presented demonstrating that NS systems comprised of buckled beams in parallel with positive stiffness springs can be used to construct quasi-zero stiffness vibration isolation systems, which provide high static but low dynamic stiffness for compact base isolation design. Transmissibility measurements of these same systems show that the nonlinearity of NS systems constructed from buckled beam structures enable tunable vibration isolation behavior and isolation from impact. Finally, modeling results will be presented demonstrating that sub-wavelength NS elements embedded in a viscoelastic material can be used to design vibration damping treatments with increased loss factor and minimally reduced stiffness to reduce the ringdown time for an impulsively loaded multi-layered beam. [Work supported by DARPA, ARO, and NSF.]

3:05

4pSA2. Periodically distributed piezoelectric patches optimization for waves attenuation and vibrations damping. Manuel Collet, Yu Fan, and Mohamed Ichchou (Dynamic of Complex systems, CNRS LTDS, Ecole Centrale de Lyon, 36 av G. de Collongue, Ecully 69131, France, manuel.collet@ec-lyon.fr)

The deformation of a structure can be understood as a superposition of waves which are induced by the excitation and reflected by the boundaries. According to this regard, attenuations of waves can lead to a strong reduction of the structural response and prevent energy to be propagated. In this work we consider periodically distributed piezoelectric patches onto the host structure so that by designing shunted electric circuits the properties of the waves can be modified. The success of the idea is also directly related to the extent of electromechanical coupling. In terms of structural modes, the coupling factor can be estimated by the open-circuit (OC) and short-circuit (SC) natural frequencies. However, in terms of waves, few criteria are available. In this work, a wave-based criterion is proposed to evaluate the coupling factor of the piezoelectric composite. To do this, enhanced Wave and Finite Element Method (WFEM) is employed to obtain the dispersion relations and the shapes of the waves. Then, the factor can be calculated in three different but equivalent formulas. An example is given thereafter, where a piezoelectric waveguide with semi-active circuits is used to control the energy flow from a source to the far-field. We show that the coupling factor is frequency dependent and it is strongly related to the geometric parameters; therefore, it significantly changes the optimal performance of the piezoelectric waveguide and it capacity to dampen vibrations.

3:25

4pSA3. Managing property distribution errors in arrays of coupled resonators. Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Andrew J. Kurdila (Mech. Eng., Virginia Polytechnic and State Univ., Blackburg, VA), John Sterling (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD), John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC), Aldo A. Glean (CertainTeed Corp., Northboro, MA), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

Arrays of small attachments can be designed to draw mechanical energy from a primary structure in a manner far in excess of their proportional size. Earlier work has shown that slight variations or errors in the property distributions of arrays of coupled resonators can have a dramatic effect on the response of primary structure to an external force. This work investigates the use of an electro-mechanical approach to making small adjustments in the stiffness of the individual elements of the array to alter the response of the primary. The electro-mechanical coupling is achieved by way of laminated thin piezoactuators mounted on a fraction of the subordinate elements. The piezoactuators are electrically coupled to a switching network that changes the effective stiffness of the subordinate elements. This ability to adjust the stiffness distribution facilitates real time control of the rate at which the energy is transferred into the coupled array. This apparent damping can then be adjusted to draw or reject energy from specific frequency bands. The presentation will describe the underling theory, present numerical results, and some preliminary experimental results.

3:45

4pSA4. Removable damping treatments for use in sheet metal fabrication. Kenneth A. Cunefare (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Many industrial fabrication processes on metallic structures generate a great deal of work place noise. For example, riveting and chiseling on light-weight, stiff aerospace structures such as fusalage components and flight surfaces subject the workpiece to repeated impacts and consequent noise generation. Many such structures may be lightly damped. Many damping treatments are targeted toward permanent installation on a structure, but if the treatments are considered to be removable, one has design freedoms including placement on a structure, as well as mass and damping means, that might not otherwise exist if the treatment had to remain in place. A shot-mass or particle-impact damper configured in an elastic, cellular array is shown to provide high loss factors and mass loading, which, along with ease of placement and removal, may provide significant reduction in component vibration during fabrication and consequent reduction in noise in the work place. The concept applied to an example aircraft panel when subjected to riveting operations yields more than 10 dB of noise reduction.

Contributed Papers

4:05

4pSA5. Solid-liner suppressor design, construction, and development. Ryan Salmon and Kenneth A. Cunefare (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, rsalmon3@gatech.edu)

Noise in a fluid system can be treated with a prototypical liner-style suppressor, an expansion chamber which includes an internal annulus of syntactic foam. A syntactic foam liner consists of host material with hollow microspheres which collapse under pressure to add compliance to the suppressor. The liner effectively increases the transmission loss of the suppressor, or ratio between inlet and outlet acoustic energy. Currently, liner-style suppressors are not commercially available. This study will investigate the integration of solid liner material within suppressor shells while also analyzing the effect of flow-smoothing diffusors on the transmission loss of the suppressor. The diffusors function to center the liner within the device, while reducing the potential for turbulence-induced self-noise. The diffusor may also impact the longevity of the liner, by reducing mechanical erosion. The results of the study should provide additional insight to the commercial viability of the liner-style suppressor.

4:20

4pSA6. Investigation of hysteresis friction in elements under complex stress state. Smirnov Vladimir, Ilya Tsukernikov, Igor Shubin, and Nina Umniakova (NIISF RAABS, Jaroslavskoe shosse, 26, Moscow 129337, Russian Federation, belohvost@list.ru)

Recent studies have indicated that the best type of damping in vibration isolation system is hysteresis (internal) damping, which effectively reduces vibration amplitudes at resonance, and it does not increase as compared with viscous damping vibration amplitudes after the resonance. Modern damping investigation methods are based on the experimental determination of the loss factor for a certain form of vibration isolators with fixed dimensions and loading parameters. At the same time, with the emergence of complex nonlinear vibration isolators, such as discussed by the authors, there is a task of expanding this theory on the case of complex designed vibration isolators. In this paper, we consider the problem of calculating the loss factor in the beam—columns of variable cross-section under complex loading conditions. A method for calculating the amount of losses in the nonlinear vibration isolators is based on Panovko's energy theory, which consists of experimental loss factor determination in the material of the vibration isolator and subsequent calculation of the loss factor in the isolator in view of obtained data.

Session 4pSCa

Speech Communication and Signal Processing in Acoustics: Advancing Methods for Analyzing Dialect Variation

Cynthia G. Clopper, Chair
Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210

Chair's Introduction—2:00

Invited Papers

2:05

4pSCa1. Reconceptualizing the vowel space in analyzing regional variation. Robert A. Fox and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Vowel space area calculated on the basis of the corner vowels has emerged as a metric for the study of regional variation, speech intelligibility, and speech development. We verify the basic assumptions underlying both the concept of the vowel space and the utility of the vowel space area in making speaker, dialect, or language comparisons. Undeniably, the traditional vowel triangle and vowel quadrilateral both fail as a metric in the context of dialect variation because substantial parts of the actual working space are excluded from analysis. Utilizing the formant values at a number of different locations for a wider range of individual vowels has significant implications for the size and shape of the resulting vowel space. Indeed, dialectal variations in vowel production can best be characterized in terms of formant density regions in the formant space and not as locations of individual vowel categories. The formant density approach is based on the assumption that vowel sounds are dynamically changing multidimensional units, which naturally overlap in the acoustic space. The formant density approach is able to minimize the amount of empty space within the overall shape while still respecting the outer boundary of the dataset.

2:25

4pSCa2. Eliciting comparable, natural speech from children and adults. Elizabeth A. McCullough (Psych., Ohio State Univ., 1835 Neil Ave., Columbus, OH 43210, eam@ling.ohio-state.edu), Cynthia G. Clopper (Linguist, Ohio State Univ., Columbus, OH), and Laura Wagner (Psych., Ohio State Univ., Columbus, OH)

In sociophonetic studies, the goal of eliciting natural speech is often at odds with the goal of eliciting speech that is easily comparable across participants. In this examination of regional pronunciation variation among English-speaking children and adults in the United States, these goals were balanced using two speech elicitation methods: color naming and picture-prompted storytelling. In the color naming task, participants saw solid blocks of color on a computer screen and were recorded saying the name of each color. In the storytelling task, participants were recorded narrating the familiar stories of *Little Red Riding Hood* and *Goldilocks and the Three Bears* using self-paced picture prompts. These methods were successfully used with participants of a wide range of ages, from pre-literate children (4 years old) to older adults (75+ years old). The color naming task yielded largely identical target words across participants because only common colors were presented. While the storytelling task allowed participants to be more creative, targets such as character names, words for important objects in the stories, and canonical lines of dialog were repeated frequently across participants. Thus, these elicitation methods avoided naturalness concerns associated with read speech while providing identical word-length and similar sentence-length examples for comparison.

2:45

4pSCa3. United States accents compared: The relation of acoustic distances and perceptual tasks. Clelia R. LaMonica (Dept. of English, Stockholm Univ., Stockholm 106 91, Sweden, clelia.lamonica@english.su.se)

This project investigates correlations between accent production and perception by comparing two sets of data: acoustic distances of American dialectal speech samples, measured using Euclidean distances of each vocalic stimuli's first three formants across a trajectory, and perceptual data in the form of online survey material and EEG tests which judge the differences between the same varieties. The speech samples used consist of six sentences from eight regions of the United States; each sentence containing phonological features that may be marked as perceptually relevant for dialect classification. Here, we examine the preliminary outcomes from such perceptual tests performed by naïve listeners, which include free classification, identification, perceived difference and ranking similarity, as well as attitude judgment tasks. Additionally, EEG tests were carried out to evaluate the relation between acoustic distance of accents and ERPs. Of special interest here is the relation between a non-regionally specified American accent and others. This "standardized" variety is judged in comparison with each of the regional accents in order to investigate listeners' perceptions of non-regional vs. regional accents, and in turn the correlation between measured accent (dis)similarity and perception.

3:05

4pSCa4. Using ultrasound articulatory signals to investigate the phonetic motivations of English /æ/ tensing. Jeff Mielke, Erik R. Thomas (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu), and Christopher Carignan (The MARCS Inst., Univ. of Western Sydney, Penrith, NSW, Australia)

A common simplifying technique in ultrasound studies of variation is to select a single representative frame for each token, sacrificing dynamic information that is often critical for understanding the phonetic motivations of phonological phenomena. We examine the phonetic motivations for tongue body raising in English /æ/ tensing (e.g., Labov *et al*. 2005) in 23 North American English speakers using phonetically meaningful time-varying articulatory signals extracted directly from ultrasound video. An articulatory measure of /æ/ tenseness is generated using regression to find the linear combination of articulatory principal components (found using EigenTongue Feature Extraction; Hueber *et al*. 2007) that best accounts for the F2-F1 difference in front vowels. We have previously shown (Carignan *et al*. 2015) that tensing before /m n/ involves tongue body raising that is timed to the vowel nucleus, whereas tensing before /g ŋ/ involves anticipating the velar closure to different degrees in different dialects. Here, we examine the phonetic motivations for this /æ/ tensing. An articulatory measure of velar fronting shows increased /g/ fronting in speakers who tense /æ/ before /g/, and a purely lingual analog of F1 shows that the effect of nasalization on F1 is much smaller than the phonological effect of pre-nasal /æ/ tensing.

3:25

4pSCa5. Modeling consonant-context effects in dialectal variation in a large database of spontaneous speech recordings. Michael Kiefte (Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca) and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, AB, Canada)

Given recent interest in the analysis of naturally produced spontaneous speech, we collected, processed, and analyzed a large database of speech samples from the Canadian province of Nova Scotia with the primary aim of examining regional variation in vowel-formant patterns. Although the actual collection of audio recordings is relatively straightforward, the analysis of spontaneous speech suffers from several disadvantages relative to that of laboratory, citation speech: Different vowels have different frequencies of occurrence, surrounding consonants have a large influence on formant peak frequencies, and the distribution of consonant contexts across different vowels is highly unbalanced. To overcome these problems, we developed a statistical procedure inspired by that of Broad and Clermont [1987, J. Acoust. Soc. Am., 81, 155] to estimate the specific effects of both onset and coda consonant-context effects on vowel formant frequencies. However, in contrast to their procedure, both vowel formant frequencies and consonant-context effects were allowed to vary freely across the duration of the vocalic portion of a syllable. Thirty-five hours of recorded speech samples from 223 speakers were automatically segmented and formant-frequency values were calculated for all stressed vowels in the database. Consonant effects were factored out to produce context-independent vowel-formant frequencies that varied across time. These data can then be used to examine dialectal variation in vowel production throughout the region.

3:45

4pSCa6. Voices of coastal Georgia. Margaret E. Renwick and Rachel M. Olsen (Linguist Program, Univ. of Georgia, University of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu)

The Linguistic Atlas of the Gulf States (LAGS; http://www.lap.uga.edu/Site/LAGS.html) contains sociolinguistic interview data from 914 speakers collected from 1968 to 1983. Impressionistic transcriptions of single words and phrases contributed to the dialectal description and mapping of the southern United States, but without systematic acoustic analysis. We gather recordings of target lexical items from ten LAGS speakers in Georgia's coastal region, focusing on data collected near St. Marys in four counties (Camden, Glynn, Charlton, and Ware). When interviewed in 1972, speakers ranged from 23 to 80 years (mean 63.7; 5M, 5F); the data thus represent dialect features of the early-mid-20th Century, including the Gullah-speaking community of St. Simons Island. The analysis focuses on features known to characterize the dialect of this region: the monophthongal vowel space, the degree of vowel diphthongization, vowel mergers before /l/ and nasals, and rhoticity of the speaker's dialect. Acoustic measurements are automatically extracted for comparison across speakers. The recordings (1 to 9 hours/speaker) were digitized as .wav within the Linguistic Atlas Project (LAP; Kretzschmar 2011). Recordings collected for the LAP are an under-attended resource that will serve as a valuable comparison to contemporary studies of regional variation in the southern United States.

Session 4pSCb

Speech Communication: Contributions to the Special Sessions (Poster Session)

Cynthia G. Clopper, Cochair
Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210

Mary E. Beckman, Cochair
Linguistics, Ohio State University, 222 Oxley Hall, 1712 Neil Ave, Columbus, OH 43210-1298

Authors will be at their posters from 4:05 p.m. to 5:00 p.m. To allow authors an opportunity to view other posters in their session, all posters will be on display from 2:00 p.m. to 5:00 p.m.

Contributed Papers

4pSCb1. How speaker identity interacts with perceptual judgments in children with residual sound errors. Sarah Hamilton, Suzanne E. Boyce, Noah Silbert, and Kirsten Mosko (Dept. of Comm. Sci. and Disord., Univ. of Cincinnati, Mail Location 379 University of Cincinnati, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu)

Recent research suggests that listeners store complex phonetic representations when learning speech. Encoding fine perceptual details, such as indexical features of the talker's voice, appears to influence performance in a variety of ways (e.g., processing speed and intelligibility of words in noise). Children with residual sound errors (RSE) for /r/ have been shown to have difficulty judging productions of /r/ from other child speakers along a normalized continuum of third formant values. We hypothesized that children with RSE may make more accurate judgments if they are given stimuli with more familiar indexical characteristics (i.e., their own speech) along the same continuum. In this study, we presented 15 children with a range of stimuli recorded from their own productions as well as productions from other children. In a forced-choice task, children indicated if the word contained a "correct" /r/. Responses to stimuli were compared across children. Initial results suggest that for RSE children, hearing one's own speech does not improve accuracy in judging the correctness of sounds in words.

4pSCb2. Using automatic alignment on child speech: Directions for improvement. Thea Knowles (Univ. of Western ON, Elborn College, Rm. 1510, School of Commun. Sci. and Disord., London, ON, Canada, thea.knowles@gmail.com), Meghan Clayards, Morgan Sonderegger, Michael Wagner, Kris Onishi, and Aparna Nadig (McGill Univ., Montreal, QC, Canada)

Phonetic analysis is labor intensive, limiting the amount of data that can be considered. Automated techniques (e.g., forced alignment based on Automatic Speech Recognition, ASR) have recently emerged allowing for larger-scale analysis. While forced alignment can be accurate for adult speech (e.g., Yuan & Liberman, 2009), ASR techniques remain a challenge for child speech (Benzeghiba *et al.*, 2007). We used a trainable forced aligner (Gorman *et al.*, 2011) to examine the effect of four factors on alignment accuracy with child speech: (1) Datasets CHILDES (McWhinney, 2000):—Spontaneous speech (single child)—Picture naming (multiple children, Paidologos data); (2) Phonetic Transcription—Manual—Automatic—CMU dictionary (Weide, 1998); (3) Training data—Adult lab data—one dataset of child data—All child data—Child & adult lab data; (4) Segment—voiceless stops—voiceless sibilants—vowels Automatically generated alignments were compared to hand segmentations. While there were limits on accuracy, in general, better results were obtained with (1) picture naming, (2) manual phonetic transcription, (3) training data including child speech, and (4) voiceless stops. These four factors increase the utility of analyzing children's speech production using forced alignment, potentially allowing researchers to conduct larger-scale studies that would not otherwise be feasible.

4pSCb3. The differential development of vowel context effects on sibilant fricatives. Patrick Reidy (Commun. Sci. & Disord., Univ. Wisconsin–Madison, 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

Previous work has found that vowel-context effects on the static acoustic properties of sibilant fricatives weaken as children age. The current study extended this prior work by analyzing the development of context effects on the spectral dynamics of English sibilants. Native adults and children (2 through 5 years old) produced /s, ʃ/ in a range of pre-vocalic contexts (/i, e, ɑ, o, u/). Effects of vowel rounding and vowel height were investigated through two pyschoacoustic measures computed from auditory spectra: peak ERB number and excitation drop (difference between maximum high-band and minimum low-band excitation). These measures were estimated from 17 20-ms windows spaced evenly across each production. Effects of vowel context on the intercept and shape of the resulting 17-point trajectories were analyzed with polynomial growth-curve models. Context effects were found to differentially weaken or strengthen in children depending on whether it was the intercept or the shape of the trajectory that was affected. For both sibilants, rounding and height effects on the peak-ERB and excitation-drop intercepts generally weakened with age; whereas, height effects on the shape factors of both trajectories tended to strengthen with age. Future work will extend the analysis to Japanese /s, ɕ/.

4pSCb4. Working memory problems of the elderly arise in the central processor, not the phonological loop. Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, nittrouer.1@osu.edu) and Joanna H. Lowenstein (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Declines in cognitive and communicative capacities can negatively impact quality of life for even healthy aging individuals. One such capacity involves working memory, which is modeled as consisting of a phonological loop that recovers phonological structure for storage and a central executive that processes that structure. In this study, working memory was investigated to evaluate which component is responsible for age-related declines. Two groups of listeners with good hearing participated: 20 young adults (18 to 32 years) and 25 elderly adults (60 to 80 years). Accuracy and speed of recall were measured for forward and backward digits, and for three sets of CVC words: nonrhyming nouns, rhyming nouns, and nonrhyming adjectives. Phonological awareness was also assessed. Results showed no differences in digit span, but significant age-related differences in accuracy of

word recall and in speed of recall for both digits and words. Phonological awareness did not differ across groups. When speed of recall was used as a covariate, effect size of age on word recall diminished. It was concluded that the problems observed in cognitive and communicative functioning of the elderly can be attributed to a slowing of processing, not to a diminishment in sensitivity to phonological structure. [NIH R01-DC000633.]

4pSCb5. Speech production in the later years: Changes in fundamental frequency and speech breathing. Eric J. Hunter, Simone Graetzer (Dept. of Communicative Sci., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu), and Ethan J. Hunter (Haslett Middle School, Haslett, MI)

With age, vocal quality can be affected by changes in articulatory, velopharyngeal, laryngeal, and breathing function. For example, past studies have shown that the voice is sensitive to hormonal changes. Other studies have indicated that there is atrophy of the vocal folds and a hardening of the laryngeal cartilages in old age. In the current study, recordings of three female adults (spanning 18 to 30 years) and three male adults (spanning 38 to 48 years, with one individual being 98 years old at the latest recording) were acquired and analyzed for changes in speech production. For a representative segment of each recording, the temporal boundaries of speech breath groups were identified by raters so that the durations of groups could be calculated. Additionally, the effect of age on speech fundamental frequency was considered. Finally, how fundamental frequency changes during a long speech task was examined, to track vocal fatigue. Results suggested a decrease in breath group duration for most subjects as age increased. Importantly, the effect of age on fundamental frequency seemed to change more than is commonly discussed (based on cross-sectional studies). Finally, there was a gender difference in how both breathing and fundamental frequency changed with age.

4pSCb6. Using vowel trajectories for southern U.S. monophthongization. Paul E. Reed (Linguist, Univ. of South Carolina, 909 Welsh Humanities Bldg., Columbia, SC 29208, reedpe@email.sc.edu)

It is widely known that monophthongization of the diphthong /ɑɪ/ is a feature of Southern U.S. English (e.g., Labov, Ash, and Boberg 2006). However, most studies of this phenomenon only use two measurement points, one from the onset and one from the glide, typically around 25% and 75% of the token's duration. While informative about whether or not a particular vocoid is monophthongal, this type of measure does not permit distinguishing among differing types of monophthongal realizations. Research including varieties from the Southern US (cf. Thomas 2000) has found that the trajectory of the vocoid can differentiate social groups. In this investigation, I track the monophthongal realizations from 24 speakers (12 men, 12 women) from Northeast Tennessee, balanced for age and education. I measure each token at 10ms intervals for both F1 and F2 for the entire vocalic duration. Plotting these measures allows for visualization and comparison of the entire trajectory of the articulation. In preliminary work in this population, flatter trajectories inversely correlate with social factors such as age, education, and local attachment. More finely nuanced measures, such as the whole vocalic trajectory, can capture social variation that may be lost when only comparing a few measurement points.

THURSDAY AFTERNOON, 5 NOVEMBER 2015 — CITY TERRACE 7, 1:30 P.M. TO 3:15 P.M.

Session 4pSP

Signal Processing in Acoustics: Algorithm, Analysis, and Beamforming

John R. Buck, Chair
ECE, UMass Dartmouth, 285 Old Westport Rd, North Dartmouth, MA 02747

Contributed Papers

1:30

4pSP1. Frequency-sum beamforming in a random scattering environment. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington 1501 NE Boat St., Seattle, Washington 98195, shimah@ldeo.columbia.edu), David C. Leckta (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Karla Mercado, Kevin J. Haworth (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

In a uniform environment, sound propagation direction(s) or the location of a sound source may be determined from array-recorded signals by beamforming. However, the beamforming results may be degraded when there is random scattering between the source and the receivers. Such sensitivity to mild scattering may be altered through use of an unconventional beamforming technique that manufactures higher frequency information from lower-frequency signal components via a quadratic product of complex signal amplitudes. This presentation will describe frequency-sum beamforming, and then illustrate it with simulation results and near-field acoustic experiments. The simulations suggest that frequency-sum beamforming may be beneficial when there is one loud source and the environment provides one primary propagation path. The experiments were conducted in either a 1.0-m-deep 1.07-m-diameter cylindrical water tank using 50 kHz and 100 kHz signals broadcast from a single source to an array of 16 hydrophones when discrete scatterers are present and absent from the tank or in a tissue-mimicking phantom with a dominant scatterer embedded and insonified at 2 MHz and the scatter received by a 128-element array. The results from frequency-sum beamforming are compared to the output of conventional delay-and-sum beamforming and minimum variance beamforming. [Work supported by NAVSEA through the NEEC.]

1:45

4pSP2. Robust plane-wave decomposition of spherical microphone array recordings for binaural sound reproduction. David L. Alon, Jonathan Sheaffer, and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, P.O.B. 653, Beer-Sheva 84105, Israel, davidalo@post.bgu.ac.il)

Rendering binaural signals from spherical microphone recordings is becoming an increasingly popular approach, with applications in telecommunications, virtual acoustics, hearing science, and entertainment. Such

binaural signals can be generated from a plane-wave decomposition of a sound field measured by a spherical microphone array. This process may exhibit ill-conditioned transformations when performed at low frequencies and using high spherical-harmonics orders, thus resulting in a poor robustness to measurement inaccuracies and noise. Previous studies have addressed this issue by employing standard regularization techniques, such as diagonal loading and radial filter limiting. In this paper, we propose an optimal frequency-dependent regularization method that balances system robustness to measurement noise against accuracy of plane-wave decomposition. Unlike previously suggested approaches, the proposed method analytically relates the measured signal-to-noise ratio to the corresponding regularization parameters, hence facilitating a means for controlling the regularization process using a closed-form expression. The method is compared to previously suggested regularization techniques in terms of spatial, temporal, and spectral effects on the resulting binaural signals. Objective results, which illustrate the improved performance of the proposed method, are complemented with a subjective validation of the regularized signals.

2:00

4pSP3. Double zero minimum variance distortionless response beamformer. Saurav R. Tuladhar and John R. Buck (ECE Dept., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, johnbuck@ieee.org)

Adaptive beamformers (ABFs) place deep beampattern notches near interferers to suppress the interferers' power in the ABF output. The common sample matrix inversion (SMI) Minimum Variance Distortionless Response (MVDR) ABF computes the beamformer weights by substituting the sample covariance matrix (SCM) for the unknown ensemble covariance matrix in the MVDR expression. Errors in the SCM estimate of interferer direction due to limited sample support or interferer motion degrade the ABFs ability to suppress the interferer. This presentation exploits array polynomial properties to design a robust ABF. The array polynomial for a uniform linear array beamformer is the z-transform of the array weights. The array polynomial zeros on the unit circle correspond to the beampattern nulls. The proposed double zero (DZ) MVDR ABF solves for the MVDR weights using the SCM for a half-aperture subarray, then convolves the half-aperture weights with themselves to obtain the full aperture ABF weights. The resulting array polynomial for the DZ MVDR ABF has second-order zeros, producing broader and deeper notches in the interferer direction. The DZ MVDR ABF outperforms the SMI MVDR and covariance matrix tapered ABFs in simulations with stationary and moving interferers. [Work supported by ONR 321US.]

2:15

4pSP4. Algorithms for measuring periodicity in F0 tracking. Xiao Chen, Hao Zhang, and Stephen A. Zahorian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, 4400 Vestal Parkway East, Binghamton, NY 13902, xchen49@binghamton.edu)

Measuring periodicity is an important measurement in speech processing. It can be used in many areas, especially for tracking fundamental frequency (F0), typically referred to as pitch. This seemingly easy measurement is made difficult since even voiced sections of speech are only semi-periodic or periodic over short intervals. In this paper, four functions for measuring periodicity are compared for both time domain and frequency domain processing. They are autocorrelation, normalized cross correlation function, spectral harmonics correlation, and normalized spectral harmonics correlation. In some cases, these four functions behave very similarly to each other; however, there are advantages and disadvantages, depending on conditions. The functions were experimentally evaluated in an F0 tracking experiment based on the Keele database, which has "ground truth" for pitch.

2:30

4pSP5. Automated detection of honeybee begging signals from long term vibration monitoring of honeybee hives. Michael-Thomas Ramsey, Martin Bencsik, and Michael Newton (Sci. and Technol., Nottingham Trent Univ., Phys. and Mathematical Sciences' College of Arts & Sci., Nottingham Trent Univ., Clifton Ln., Nottingham NG11 8NS, United Kingdom, n0530709@ntu.ac.uk)

We are using high performance accelerometers embedded in the honeycomb to create long term vibrational data sets comprising a range of individual bee pulsed vibrational messages. The statistics of the "vibrational language" of the honeybee can thus be explored. The aim of this study is to create and optimize software that detects and analyses the occurrences of honeybee "begging signals." Recording from a beehive were monitored for 117 days (July to November 2014) from within its central frame using two ultra-high performance accelerometers (Brüel and Kjær, 1000 mV/g), one in the center and the other one 7 cm lower down. Home-build Linux 'bash code' was written to continuously record the accelerometer output into separate one-hour long files, without any data loss. Matlab® code has been developed, that compares a template and instantaneous spectrograms to detect a begging signal. Six independent parameters of the template signal waveform were further optimized, in order to yield the maximum correct detections. Finally, the detection threshold was tuned ensuring that only begging signals were detected. The results show long term trends in the begging signal statistics, a breakthrough in exploring seasonal variations in the honeybee vibrational language.

2:45

4pSP6. Time varying broadband acoustic response compensation at very low signal to noise ratio. Praswish Maharjan and Paul J. Gendron (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, pmaharjan@umassd.edu)

Shallow water ocean acoustic environments are challenging due to their multi-path arrival structure and the various Doppler processes that are associated with platform motion and surface interactions. For moving platforms, each acoustic arrival exhibits a Doppler offset that can be quite significant, which results in an overall response that is doubly spread, one which disperses signals in both time and frequency. A virtual ocean acoustic laboratory based on the Bellhop ray tracing model is used to study the distortion of broadband waveforms through dynamic shallow water ocean acoustic environments. Acoustic response functions are presented in order to illuminate the challenges posed by shallow water environments and mobile platforms. The shared Doppler process of the arrivals is illustrated in order to illuminate the value of bulk Doppler compensation schemes. This virtual acoustic laboratory is used to test spread spectrum communication performance at very low SNR. Bulk Doppler compensation schemes derived from joint estimation of symbols and the acoustic response are compared to the actual path dilation processes. Simulation results are presented for fixed source with moving receiver shallow water environments at diverse center frequencies and bandwidths.

3:00

4pSP7. Wind farm infrasound—Are we measuring what is actually there or something else? Steven E. Cooper (The Acoust. Group, 22 fred St., Lilyfield, NSW 2040, Namibia, drnoise@acoustics.com.au)

In the olden days of acoustics (pre digital), low frequency analysis used analog narrow band filters and cathode ray oscilloscopes for special problems leading to the general use of peak values. Analog filters have time constants that can affect the derived rms values requiring caution where high crest factors are involved. Modern narrowband digital analysis is based on a FFT of the time signal to extract the periodic function that occurs in the time domain that are then displayed as discrete peaks in the frequency domain. FFT analysis of turbines show discrete infrasound peaks at multiples of the blade pass frequency in addition to sidebands in the low frequency range spaced at multiples of the blade pass frequency. Are these signals actually there or are they a product of modern day analysis. Is the infrasound signature a clue to a different area of investigation? The paper will show the results of testing to compare old fashioned and modern day analysis.

Session 4pUW

Underwater Acoustics and Signal Processing in Acoustics: Environmental Variability Impact on Shallow Water Acoustics II

Kevin D. Heaney, Cochair
OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039

Sergio Jesus, Cochair
University of Algarve, Campus de Gambelas, Faro 8005-139, Portugal

Contributed Papers

1:00

4pUW1. Remote sensing for bottom inversion. Sergio M. Jesus (Univ. of Algarve, Campus de Gambelas, Faro 8005-139, Portugal, sjesus@ualg.pt)

The scenarios of interest for estimating bottom and sub-bottom physical properties now encompass both deep and shallow, or very shallow, coastal waters, for the deployment of renewable energy platforms (e.g., wind farms and wave/tidal energy plants). This new paradigm, together with the continuous requirement for reducing survey time (and cost), spinned out the concept of a distributed and reconfigurable seismic survey system composed of a fleet of autonomous underwater vehicles (AUV) carrying acoustic sensing arrays. Such system poses a number of technical as well as scientific challenges, among which that of sensor positioning for optimal bottom inversion performance in a given scenario. The present work addresses this issue through the eye of the sensor configuration that maximizes diversity and proposes sampling incoherence bounds for 1, 2, and 3D array systems. Random sampling is a concept that favors diversity and allows for the usage of low-complexity inversion schemes such as those based on compressed sensing. Simulations on realistic environments illustrate the proposed concept. [This work is part of project WiMUST—Widely Scalable Mobile Underwater Sonar Technology funded under program H2020 of the European Union.]

1:15

4pUW2. Source depth discrimination: An evaluation and comparison of several classifiers. Ewen Conan, Julien Bonnel (Lab-STICC, ENSTA Bretagne, 2 rue François Vermy, Brest 29200, France, ewen.conan@ensta-bretagne.org), Thierry Chonavel (Lab-STICC, Telecom Bretagne, Brest, France), and Barbara Nicolas (Créatis, Université de Lyon, Lyon, France)

Source depth estimation with a vertical linear array generally involves mode filtering, followed by matched-mode processing. However, this method has two main limitations: the problem of mode filtering is ill-posed in the case of partially spanning arrays; matched-mode processing is sensitive to environmental mismatch. Therefore, concerns for robustness motivate a simpler approach. The problem of depth estimation is reduced to a binary classification problem: source depth discrimination. Its aim is to evaluate whether the source is near the surface or submerged. These two hypotheses are formulated in terms of normal modes, using the concept of trapped and free modes. Several classification rules, based on modal filtering or on subspace projections, are studied. Monte-Carlo methods are used to evaluate their performance and compute receiver operating characteristics. This allows the choice of a discrimination threshold according to some expected performance. The benefits of considering a source depth discrimination problem rather than a source localization one are highlighted. The influence of noise and environmental mismatch are investigated, as well as the choice of the discrimination depth and the choice of the limit between trapped and free modes.

1:30

4pUW3. Interpretations of the frequency difference autoproduct in multipath environments. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 2010 Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

When locating remote acoustic sources in a shallow ocean sound channel, the established array signal processing technique known as matched field processing (MFP) has shown much success. However, MFP is sensitive to mismatch between the modeled and actual environments, and may fail to localize acoustic sources in the presence of such mismatch, particularly at high frequencies. A recent nonlinear array signal processing technique, frequency difference MFP (*Abadi et. al.* 2012, *Worthmann, et. al.,* under review), has shown some success in localizing high frequency sources by moving the replica calculations to a lower, out-of-band, difference frequency where the detrimental effects of environmental mismatch are less severe. To extract the requisite out-of-band difference frequency information from the measured signals, a quadratic product, termed the autoproduct, is formed from complex signal amplitudes separated by the difference frequency but still lying within the signal bandwidth. Through the use of simple multipath propagation environments, the nature of this autoproduct is explored, and the reasons that it provides out-of-band field information are presented. More complex propagation environments are simulated as well to demonstrate some of the expected and unexpected behaviors of the autoproduct. [Sponsored by the Office of Naval Research and the National Science Foundation.]

1:45

4pUW4. The relation between the waveguide invariant and array invariant. Hee-Chun Song and Chomgun Cho (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

The waveguide invariant β is based on the dependence of group speed on phase speed and summarizes the robust interference phenomenon in the range-frequency plane. Over the last decade the elegant approach has been utilized for various applications including passive source ranging. Separately, the array invariant approach [Lee and Makris, J. Acoust. Soc. Am., vol. 119, pp. 336–351 (2006)] has been proposed for a robust source-range estimator from beam-time intensity data using either a horizontal or vertical array. In this letter, it is shown that the array invariant can be derived from the waveguide invariant theory assuming $\beta = 1$.

2:00

4pUW5. Robust source-range estimation using a vertical array and the array invariant. Chomgun CHO, Hee-Chun Song, and William S. Hodgkiss (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, San Diego, CA 92093-0238, chomgun@gmail.com)

The array invariant based on beam-time migration has been proposed for instantaneous source-range estimation in acoustic waveguides using a horizontal or vertical array. With minimal knowledge of the environment, the approach has been demonstrated successfully with experimental data in shallow water using a horizontal array. Moreover, it was determined recently that the array invariant can be derived from waveguide invariant theory based on the dependence of group speed on phase speed. In this paper, the array/waveguide invariant approach to source-range estimation is applied to a vertical array in a fluctuating ocean environment over a one-day period. Specifically, the range estimates using a 12-m long vertical array in ~100 m deep water are within 9–16% relative error for a 2–3 kHz source at 6-km range, demonstrating the robustness of this approach.

2:15

4pUW6. Single and multiple snapshot compressive matched field processing. Kay L. Gemba, William S. Hodgkiss, and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Matched field processing is a generalized beamforming method which matches received array data to a dictionary of replicas to locate and track a source. The solution set generally is sparse since there are considerably fewer sources than replicas. This underdetermined problem can be solved with sparse processing (SP) which potentially is attractive for several reasons. The traditional spatial matched-filter problem is reformulated as a convex optimization problem subject to a sparsity constraint. For example, an elastic net seems to be an appropriate penalty in order to find the best match among a correlated group of replicas. Another advantage is that SP does not require inversion of the sample covariance matrix and therefore can outperform conventional high-resolution processors in snapshot deficient scenarios (i.e., fast moving sources). A third potential advantage is that SP can achieve super-resolution at high SNR and discriminate between closely spaced sources. Here, we demonstrate the performance of single and multi-snapshot SP to track a towed source using the SWellEx-96 data set. Results are benchmarked using the Bartlett and the white noise constraint processors. We further discuss the processing of multiple frequencies in order to improve the source tracking.

2:30

4pUW7. The exponential decay of underwater acoustic intensity with increasing altitude of low-frequency sound from a Robinson R44 helicopter. Dieter A. Bevans and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, dbevans@ucsd.edu)

A series of underwater acoustic experiments utilizing a Robinson R44 helicopter and an underwater receiver station has been conducted in shallow (16 m) and deep (>100 m) water. The receiver station consisted of an 11-element nested hydrophone array with a 12 m aperture. In the shallow water experiments the array was configured as a horizontal line (HLA) 0.5 m above the seabed; whereas in deep water the array was suspended from the surface in a vertical line (VLA). An in-air microphone was located immediately above the surface. In this paper the power spectral density as a function of helicopter altitude will be reported from measurements on a single hydrophone. The main rotor blades of the helicopter produce low-frequency harmonics, the lowest frequency being ~13 Hz. The tail rotor produces a sequence of harmonics approximately six times higher in frequency. Between heights of 30 m to 600 m above the sea surface the underwater intensity was found to decay exponentially with increasing helicopter altitude. Interpretation of the observed low-frequency sound signatures is being facilitated with a numerical three-layer (atmosphere-ocean-sediment) acoustic propagation model in which the source may be moving or stationary. [Research supported by ONR, NAVAIR, and SIO.]

2:45

4pUW8. The effect of signal bandwidth on partially saturated broadband ocean acoustic transmission scintillation in a shallow water waveguide. Delin Wang, Wei Huang, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., 302 Stearns Ctr., Rm. 312, 360 Huntington Ave., Boston, MA 02115, wang.del@husky.neu.edu)

The acoustic field in a shallow water waveguide at small source-receiver separations <3 km are partially saturated. Here, we examine the scintillation statistics of partially saturated broadband ocean acoustic transmissions in the low to mid frequency range collected using a high-resolution towed horizontal coherent hydrophone array system. The standard deviation of intensity fluctuations and scintillation index are quantified as a function of signal bandwidth and source-receiver separation. The amplitude and intensity distributions are compared with corresponding distribution models for the partially saturated field to determine the number of independent statistical fluctuations present in the broadband data.

3:00–3:15 Break

3:15

4pUW9. Three-dimensional inversion technique in ocean and seabed acoustics using the parabolic equation method. Camilo C. Roa (Ocean and Mech. Eng., Florida Atlantic Univ., 101 North Beach Rd., Dania Beach, FL 33004, croa1@fau.edu) and George V. Frisk (Ocean and Mech. Eng., Florida Atlantic Univ., Boca Raton, FL)

A three-dimensional parabolic equation (PE) and perturbation approach is used to invert for the range-dependent geoacoustic characteristics of the seabed. The model assumes that the sound speed profile is the superposition of a known range-independent profile and an unknown depth and range-dependent perturbation. Using a Green's function approach, the total measured pressure field in the water column is decomposed into a background field, which is due to the range-independent profile, and a scattered field, which is due to the range-dependent perturbation. When the Born approximation is applied to the resulting integral equation, it can be solved for the range-dependent profile using linear inverse theory. For simplicity, the sound speed profile in the water column is assumed to be known, and the range-dependent perturbation is added to the index of refraction $n(x,y,z)$, rather than the sound speed profile $c(x,y,z)$. The method is implemented in both Cartesian (x,y,z) and cylindrical (r,θ,z) coordinates with the forward propagation of the field in x and r, respectively. Synthetic and experimental data are used to demonstrate the validity of the method. Keywords: Three-dimensional parabolic equation method, geoacoustic inversion, range-dependent sound speed profile, linear inversion, Born approximation, and Green's functions.

3:30

4pUW10. Broadband impulse response modeling using the single frequency parabolic equation solution. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

The parabolic equation has been considered the workhorse of low to mid frequency propagation modeling in range dependent environments due to its accurate handling of both refraction and diffraction. The frequency domain solution approach of the PE has severely limited its use in broadband pulse synthesis die to the computational cost of computing each Fourier component. In this paper a hybrid approach that blends the PE forward model for field computation and an analytic model for travel time computation is presented. The PE is computed at the center frequency in a range dependent environment. The beam arrival angle for a virtual line array at the receiver depth (as well as sea surface and sea floor) is stored at each range. The average group slowness as a function of phase speed at the source (wave number invariant) is computed using analytic acoustics methods (Snell's law). The impulse response at each range is computed by incoherently adding the pier of each beam at the arrival time of the associated phase speed energy (the range averaged group slowness). This approach reproduces the behavior of the pulse spread, without the cost of the broadband PE.

3:45

4pUW11. Three-dimensional iterative parabolic approximations. Pavel S. Petrov (School of Natural Sci., Far Eastern Federal Univ., 8 Suhanova St., Vladivostok 690950, Russian Federation, petrov@poi.dvo.ru)

A hierarchy of the 3D coupled parabolic equations is derived by the method of multiple scales. The solutions of the derived equations represent the successive terms in an asymptotic expansion of the solution of the 3D Helmholtz equation. The equations are complemented with the consistent interface and boundary conditions. The Cauchy initial conditions for the parabolic equations are set up in such a way that the solution in the far field approximates the solution of the Helmholtz equation in the unbounded 3D space. The derived parabolic equations are used to solve a problem of the propagation in a perfect 3D wedge. The comparison to the image source solutions is used for the validation of the proposed parabolic approximation.

4:00

4pUW12. Stochastic-basis matched-field processing. Steven I. Finette, Peter C. Mignerey, and Roger M. Oba (U.S. Naval Res. Lab., Peter Mignerey code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil)

Matched-field processing (MFP) suffers serious degradation due to environmental mismatch between received acoustic-field vectors and modeled replica vectors. Physical reasons for degradation include uncertainty caused by incomplete descriptions of the parameters and fields necessary for correct specification of the acoustic waveguide (i.e., environmental uncertainty), and system uncertainty associated with incomplete knowledge of the array configuration, source depth, etc. Recent research in the theory of stochastic-basis expansions (polynomial chaos) provides a mathematically consistent way of incorporating both types of uncertainty into MFP. Such expansions are used efficiently to construct replica matrices that steer high-rank subspaces capable of capturing signals with uncertain wavefronts. When combined with cross-spectral density matrices, stochastic-basis steering matrices enable the design of new processors with properties not previously evaluated in a MFP context. A maximum likelihood processor is developed which incorporates environmental uncertainty through polynomial chaos expansions. The processor can be written as a Frobenius product between an estimated cross-spectral density matrix and the inverse of a stochastic-basis replica matrix. This talk will outline the theoretical foundation of stochastic-basis MFP, and illustrate the method with some simulations. [Work supported by the Office of Naval Research.]

4:15

4pUW13. Acoustic propagation in the Strait of Georgia. Nicos Pelavas, Sean Pecknold (DRDC Atlantic Res. Ctr., 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, nicos.pelavas@drdc-rddc.gc.ca), Matthew Coffin, Kevin Dunphy (GeoSpectrum Technologies, Inc., Dartmouth, NS, Canada), and Dugald Thomson (Dept. of National Defence, Trinity - Acoust. Data Anal. Ctr., , Halifax, NS, Canada)

In recent years, there has been a proliferation of Ocean Observing Systems (OOS) along with a wide distribution of their associated data products. The collected data support scientific research, industry, and government organizations by providing long term measurements of biological, chemical, and physical properties of the ocean environment. However, the collection and distribution of underwater acoustic data poses a potential security risk for naval vessels operating in the vicinity of OOS. The Canadian Forces Maritime Experimental and Test Ranges (CFMETR) provide an underwater tracking facility for naval tests, and are approximately 50 km from hydrophones of the Victoria Experimental Network Under the Sea (VENUS) observatory. Under an existing CFMETR-VENUS agreement, data are diverted during certain naval tests. In order to minimize the frequency of these data diversions, a study is being conducted to investigate acoustic propagation in the Strait of Georgia. The results of acoustic modeling and measurement of transmission loss from CFMETR to VENUS will be presented. A software application called CAVEAT is also presented. The application was developed to integrate the transmission loss results along with other sonar parameters to enable operators at CFMETR to determine the risk of acoustic exposure.

4:30

4pUW14. A comparison of the reflection coefficient predictions of two competing sediment acoustic models. Anthony L. Bonomo, Nicholas Chotiros, and Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Currently, there are several competing models that have been used to describe the acoustic properties of sandy sediments. These models include those that assume the sediment to behave as an acoustic fluid, a viscoelastic solid, and as a porous media following Biot theory. Perhaps the two most sophisticated acoustic models that have been applied to sand are the Viscous Grain Shearing (VGS) model of Buckingham and the Extended Biot (EB) model of Chotiros. While both of these models have been found to agree with measured sound speed dispersion and attenuation data, previous work has shown that the reflection coefficients predicted using these models disagree. In this work, the reflection coefficient predictions of the VGS and EB models will be compared for both the case of a homogeneous sand half-space and the case of a sand layer overlying a rock substrate. [Work supported by ONR, Ocean Acoustics.]

4:45

4pUW15. Hydroacoustic survey of geotechnical activities for the Virginia offshore wind technical advancement program. Erik J. Kalapinski (Energy Programs, Tetra Tech, Inc., 160 Federal St., Fl. 3, Boston, MA 02210, erik.kalapinski@tetratech.com) and Kristjan A. Varnik (Eng., Tetra Tech, Inc., Boston, MA)

As offshore wind energy development increases across the eastern seaboard, there is a growing need to determine the short and long-term effects of activities associated with this development on marine ecosystems. One area of particular importance is the potential effects of underwater noise on marine life. To better characterize underwater sound levels associated with geotechnical activities, a hydroacoustic measurement program of geotechnical survey activities was completed in support of Dominion's Virginia Offshore Wind Technical Advancement Program. An important component of this project was the advancement of new technologies and the use of best available science to collect data for more accurate impact determinations. The overall goal of the hydroacoustic measurement program was to field-verify the projected acoustic impacts during geotechnical activities. This new insight will support both decision making in the execution of offshore wind site characterization surveys, and reduce the potential for future impacts. Measurements were made using a combination of equipment including a cabled real-time hydroacoustic analysis systems and fixed autonomous static recorders. Upon conclusion of the hydroacoustic survey, data were downloaded and directly correlated with daily activity logs from the vessels used in performing the offshore geotechnical work thereby providing the means to coordinate acoustic events.

Poster Papers

Posters will be on display from 1:00 p.m. to 5:00 p.m. Authors will be at their posters from 3:00 p.m. to 3:15 p.m.

4pUW16. Computationally efficient algorithm for Kirchhoff approximation. Donghyeon Kim (Ocean Eng., Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, 253, Ocean Sci. and Technol., Busan ASIKRKS012BUSAN, South Korea, donghyeonkim@kmou.ac.kr), Hubum Jin (Mathematics, POSTECH, Pohang, South Korea), Yoon Hee Ji (LIG Nex1, Seongnam, South Korea), Ho Seuk Bae, Woo-Shik Kim (Agency for Defense Development, Changwon-si, South Korea), and Jea Soo Kim (Ocean Eng., Korea Maritime and Ocean Univ., Busan, South Korea)

In order to simulate the target echoes scattered from submerged underwater objects, Kirchhoff approximation is widely used for high frequency region. Since Kirchhoff integration is based on integrating the contributions from discretized boundary elements, the computation can be time-consuming especially for broadband pulses. In this study, a numerically efficient method for generating the scattered signal in time domain based on convolution is proposed and tested. It is shown that the computational time can be reduced by an order of 10–100 in typical cases.

4pUW17. Temporal variation of transmission loss by internal tide in the southern sea of Jeju island in summer. Juho Kim, Hansoo Kim, Dong-Guk Paeng (Jeju National Univ., Ara 1 dong, Jeju 064-756, South Korea, lizard@jejunu.ac.kr), and Ig-Chan Pang (Jeju National Univ., Jeju, Jeju Special Self-Governing Province, South Korea)

Temporal variations of acoustic transmission loss (TL) affected by internal tide are simulated using oceanic data measured at two sites in the southern sea of Jeju island. Temperature and salinity were measured every hour for 25 hours during July 27th and 28th 2009. The periodic fluctuation of temperature due to the internal tide was observed and its vertical displacement was more than 10m. In order to investigate the variation of TL, acoustic propagation between two measurement sites (3.8 km distance) was computed at a source depth of 10 m. Acoustic propagation model RAM was used for the simulation. Standard deviations of TL variation were 4.2 dB and 3.7 dB for center frequencies of 100 Hz and 1 kHz with 1/3 octave band, respectively. The lower frequency was more correlated with the tidal period than the higher one. Detection range was also varied by the internal tidal, up to 1 km for the 60 dB detection level. These results imply that tidal variation of TL should be considered for acoustic researches at the southern sea of Jeju island. [This work was supported by UD130007DD and IITP-2015-H8601-15-1004.]

4pUW18. Modeling of sound in coastal oceans with a finite volume method. Wen Long (PNNL, 1100 Dexter Ave North, Seattle, WA 98109, wen.long@pnnl.gov), Ki Won Jung (PNNL, Richland, WA), Zhaoqing Yang (PNNL, Seattle, WA), Zhiqun Deng (PNNL, Richland, WA), and Andrea Copping (PNNL, Seattle, WA)

In this research, the finite volume method is employed to solve the 3D Helmholtz equation of sound propagation in the coastal environment. The grid system consists of triangular grids in horizontal plane and sigma-layers in vertical dimension. A 3D sparse matrix solver with complex coefficients is formed for solving the resulting acoustic pressure field. The recent CSLP and ADEF1 preconditioning methods are applied to efficiently solve the matrix system iteratively with MPI parallelization using a high performance cluster. This model is then coupled with the Finite Volume Community Ocean Model (FVCOM) for simulating sound generated by offshore wind energy platform constructions in a range-dependent setting. Details of the development and initial validation will be presented. Keywords: Sound, Coastal Ocean, Finite Volume, Offshore Wind, Coupling, CSLP, ADEF1, Helmholtz equation.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday, 3 November

Committee	**Start Time**	**Room**
Engineering Acoustics	4:30 p.m.	Orlando
Acoustical Oceanography	7:30 p.m.	River Terrace 2
Animal Bioacoustics	7:30 p.m.	City Terrace 9
Architectural Acoustics	7:30 p.m.	Grand Ballroom 3
Physical Acoustics	7:30 p.m.	St. Johns
Psychological and Physiological Acoustics	7:30 p.m.	Grand Ballroom 7
Structural Acoustics and Vibration	7:30 p.m.	Daytona

Committees meeting on Wednesday, 4 November

Committee	**Start Time**	**Room**
Biomedical Acoustics	7:30 p.m.	Clearwater
Signal Processing in Acoustics	8:00 p.m.	City Terrace 7

Committees meeting on Thursday, 5 November

Committee	**Start Time**	**Room**
Musical Acoustics	7:30 p.m.	Grand Ballroom 1
Noise	7:30 p.m.	Grand Ballroom 2
Underwater Acoustics	7:30 p.m.	River Terrace 2

Session 5aAB

Animal Bioacoustics: New Discoveries in Bat Vocal Communication

Kirsten M. Bohn, Cochair
Florida International University, Dunning Hall, Suite 434, Baltimore, MD 21218

Michael Smotherman, Cochair
Biology, Texas A&M University, 3258 TAMU, College Station, TX 77843-3258

Chair's Introduction—8:20

Invited Papers

8:25

5aAB1. Signature calls predict foraging success in big brown bats (*Eptesicus fuscus*). Genevieve S. Wright (Dept. of Biology, Univ. of Maryland, College Park, MD 20742, myotis@gmail.com), Chen Chiu, Wei Xian (Dept. of Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Gerald S. Wilkinson (Dept. of Biology, Univ. of Maryland, College Park, MD), and Cynthia F. Moss (Dept. of Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Animals foraging in the dark must simultaneously pursue prey, avoid collisions, and interact with conspecifics, making efficient non-visual communication essential. A variety of birds and mammals emit food-associated calls that inform, attract, or repel conspecifics. While echolocation by the insectivorous, aerial-hawking big brown bat *(Eptesicus fuscus)* has been studied extensively, communicative calls used by this species have received comparatively little research attention. We report on a rich repertoire of vocalizations produced by big brown bats in a large flight room equipped with synchronized high-speed stereo video and audio recording equipment. We also provide evidence that a specific social call, the "frequency-modulated bout" (FMB), which is emitted only by males, exclusively in a foraging context, and only when conspecifics are present, predicts the caller's foraging success and is individually distinct. Bats were studied individually and in pairs, while sex and experience with a foraging task were experimentally manipulated. Individuals emitting a higher number of FMBs showed greater prey capture success. Following FMB emission, inter-bat distance, diverging flight, and the other bat's distance to the prey increased. These findings highlight the importance of vocal communication for nocturnal animals mediating interactions in a fast-paced foraging setting.

8:45

5aAB2. Ultrasonic and superfast: Design constraints on echolocation in bats. John Ratcliffe (Dept. of Biology, Univ. of Toronto Mississauga, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, j.ratcliffe@utoronto.ca)

Recent work from our group demonstrates that two exceptional characteristics of bat biosonar—bats' extremely high call emission rates and these calls' ultrasonic frequencies—reflect biomechanical constraints of the vocal apparatus. We hypothesized that smaller bats, with their smaller mouths, emit higher frequencies to achieve sufficiently directional sonar beams, and that variable directionality is critical for bats. We found that six aerial hawking, vespertilionid bat species produced sonar beams of similar shape and volume, and we predict that many bats adjust their acoustic field of view to suit habitat and task. We speculate that sonar beam shape has been an evolutionary constraint on echolocation and explains the bat size-call frequency correlation. During the terminal phase of an aerial hawking attack on an insect, bats produce a "buzz," increasing information update rates by producing >160 calls/second. We discovered that bats use specialized superfast muscles to power these rapid call rates. We also show that laryngeal motor performance, not call-echo overlap, limits maximum call rate. We suggest that the advantages of rapid auditory updates on prey movement have selected for superfast laryngeal muscle. Taken together, our results provide further evidence that bat biosonar is a dynamic sensory system, a sensory system that allows bats to adjust and optimize their acoustic fields of view and to update their auditory scene at rates >160 times/second to optimize airborne prey detection and tracking.

9:05

5aAB3. Probabilistic strategies for dynamic coordination of biosonar emissions in social groups. Michael Smotherman (Biology, Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, smotherman@tamu.edu)

Echolocating bats are social animals and must be able to use their biosonar capabilities in a wide range of social contexts. Bats roosting or flying in groups routinely adapt their pulse emissions to accommodate sharing the acoustic space with potentially many bats. In this regard, echolocation may be bound by the same rules and constraints governing social communication. Bats can alter pulse acoustics, temporal patterns, projections patterns, and locomotor trajectories to minimize acoustic interference across individuals. Here, we present evidence that the highly gregarious free-tailed bat, *Tadarida brasiliensis*, primarily relies upon probabilistic strategies for mitigating acoustic interference rather than trying to predict or extract information from the pulses of other bats. When free-tailed bats hear

5a FRI. AM

the pulses of another bat they shift the timing of subsequent pulse emissions following a randomized back-off algorithm similar to those that have evolved in artificial wireless communications systems. Importantly, this behavior is only effective if all bats strictly follow the same strategy. I will present the results of recent measurements of this back-off algorithm and use computational modeling to illustrate how and when it improves sonar imaging in social settings.

9:25

5aAB4. Vocal flexibility and regional variation in free-tailed bat songs. Israel Salazar (Biological Sci., Florida Int.. Univ., 11200 S. W. 8th St., Office DM 438A, Miami, FL 33199, isala004@fiu.edu) and Kirsten Bohn (Biology, Johns Hopkins Univ., Baltimore, MD)

Male Brazilian free tailed bats (*Tadarida brasiliensis*), sing to attract females and defend territories during the mating season. These songs are unusual among mammalian vocalizations in that they are highly complex and hierarchically structured. Songs are composed of multiple syllables which are combined into three phrases that vary in number and order across renditions. While much work has been done on regional vocal variation in birds, relatively a few studies have found similar evidence in mammals. This study aimed to determine if *T. brasiliensis* songs vary across agographical regions. To accomplish this, we compared spectro-temporal characteristics, and phrase composition of songs recorded from wild colonies in various locations throughout Florida, Georgia, and Texas. Although we found considerable inter-individual and within-individual variation, there are salient regional differences in song structure. These findings combined with a high level of within-individual flexibility, and the lack of local adaptation in foraging echolocation pulses, support a possible role of vocal learning in song production in this species.

9:45–10:00 Break

10:00

5aAB5. Geographic variation in contact calls emitted by a leaf-roosting bat suggest distinct modes of vocal transmission. Karina Montero and Erin H. Gillam (Biological Sci., North Dakota State Univ., NDSU Dept. 2715, Stevens Hall 201, PO Box 6050, Fargo, ND 58108-6050, Erin.Gillam@ndsu.edu)

Group behaviors, such as coordination and information exchange, are typically mediated by acoustic signals known as contact calls. Although these vocalizations are widespread, the mechanisms driving variation in acoustic features within and between populations remain poorly understood. Our study examines whether patterns of variation in two contact calls emitted by Spix's disk winged bats, *Thyoptera tricolor,* are congruent with patterns of genetic distance among populations isolated by a geographic barrier. *T. tricolor* is a leaf roosting bat that forms stable social groups and exhibits all-offspring philopatry. To evaluate if vocal variation between groups is influenced by genetic distance, we studied the variation in microsatellite allele frequencies at multiple sites on the Caribbean and Pacific mountain slopes. We found that the geographic variation patterns differed between the two types of calls studied, and we argue that this indicates distinct modes of vocal transmission. Our results suggest that one contact call is likely socially transmitted via vocal learning, while the congruence between patterns of genetic differentiation and acoustic variation for the second call type suggest this is an inherited trait. Further research is needed to better understand the role of vocal learning and genetic transmission of contact calls emitted by *T. tricolor*.

10:20

5aAB6. Singing away from home: Song is used to create and defend foraging territories in the African megadermatid bat, *Cardioderma cor*. Grace C. Smarsh (Biology Dept., Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, gsmarsh@bio.tamu.edu)

The diversity of song repertoires and functions of singing in mammals are not well known. In bats, singing in the roost to court and defend mates has been studied; however, the concept of territorial behaviors and the role of singing outside of the roost is poorly understood. The heart-nosed bat, *Cardioderma cor*, roosts in mixed-sex and age groups in the hollows of baobab trees, but disperses to exclusive areas whereupon they move about foraging and singing. We investigated singing in this species by mist-netting, pit-tagging, and tracking 12 singing individuals during which we recorded songs and collected singing and movement behavioral data. We conducted song playbacks to further test song function. Male *C. cor* individuals return to same foraging area nightly, which are often over 100 m across, and favor perches where they sing back and forth with neighbors. Low-frequency, repetitive syllables are likely adapted for song transmission across the cluttered bush habitat. Songs vary within and across individuals both spectrally and temporally. Song playbacks elicited aggressive responses, confirming that these bats use song as part of their territorial defense strategy, similar to the way song is used by many songbirds.

10:40

5aAB7. Describing the social behavior of the Indiana bat at day roost sites. Caroline M. Byrne (Biology, Indiana State Univ., 276 Canco Rd., Portland, ME 04103, caroline.byrne@briloon.org) and Joy M. O'Keefe (Biology, Indiana State Univ., Terre Haute, IN)

Bats are highly social, but the study of bat social behavior was limited until recently due to technological limitations. Most of bat behavior is imperceptible to our senses, including both their use of ultrasound and their nocturnal activities. During the maternity seasons (May–August) of 2013 and 2014 near Plainfield, Indiana, we recorded Indiana bat (*Myotis sodalis*) roost site behaviors with passive emergence count observation, video (Sony Nightshot HandyCams and IR lights), and acoustics (Pettersson D500X acoustic detectors). The objective of this study was to compile a catalog of visual and acoustic behaviors seen at Indiana bat day roost sites. Thus far, we have detected 29 specific types of visual behavior, and five general types of acoustic behavior. The documented behaviors include visual behaviors similar to those categorized as "checking behavior" and acoustic behaviors seen in contexts of agonistic, echolocation, infant isolation, and disturbance in little brown bats (*Myotis lucifugus*). These are some of the first systematic observations of social behavior for Indiana bats. Understanding the social behaviors of these highly social bats is crucial to gaining a full understanding of their life cycle and daily requirements.

11:00

5aAB8. Territorial calls analysis in the broad tailed bat (*Nyctinomops laticaudatus*). Fernando J. Montiel Reyes (Bioconservación y manejo, Escuela Nacional de Ciencias Biológicas, Instituto Politécnico Nacional, 60 Brezo St., Colonia Nueva Santa María, Mexico City, Azcapotzalco 02800, Mexico, fercho_mom@hotmail.com), Kirsten Bohn (Johns Hopkins Univ., Baltimore, MD), and Jorge Ortega (Bioconservación y manejo, Escuela Nacional de Ciencias Biológicas, Instituto Politécnico Nacional, Mexico City, Mexico)

Some bat families perform vocalizations called "songs" to mate, conformed by three element types: chirps, trills, and buzzes. Also, those might be territorial warning calls; but its syntax remains to be understood. Our goal is to get information about mating calls' patterns and its clarified function for a molossid bat species. We used ultrasound microphones to record vocalizations from a colony of *Nyctinomops laticaudatus*, during three periods through 2013–2014. We performed spectrogram analysis to characterize vocalizations with chirp number, duration, and acoustic frequency measurements. Results show males perform structured and complex songs. Vocal repertoire features two chirp types, trills, and buzzes as in other species. Chirp A is a short descendant frequency modulated pulse of 8.05 ms (SD = 3.53 ms); chirp B is a downward FM larger than A with an upward final segment of 43.93 ms total length (SD = 14.85 ms). Distinctively *Nyctinomops laticaudatus* performs songs of a series of three to nine compound chirps, starting with B element of high plasticity. Additionally structure is modified by the syntax, with shorter phrases when the first element is a trill or a buzz.

11:20

5aAB9. Bats as new models for social communication. Kirsten M. Bohn (Johns Hopkins Univ., Dunning Hall, Ste. 434, Baltimore, MD 21218, kbohn1@jhu.edu)

Bat echolocation has been the focus of extensive acoustic research for over 50 years. However, our knowledge of bat social communication is in its infancy, having really only begun to develop over the last decade. This is because only recently have ultrasonic recording and playback become affordable, rugged, and portable—making field research highly expedient. These advances have in essence opened up an entirely new frontier in acoustic research. Indeed a very large frontier with over 1000 diverse species, nearly all of which are highly social. Here, I review the types of vocalizations bats produce, from echolocation calls, to simple calls to complex songs. For example, simple infant isolation calls are produced by the majority of mammals with little to no specialization in bats. In contrast, our work on Molossid bats shows that they embed "echolocation" calls into complex song phrases while roosting and in flight. In this case, social communication is likely an exaptation of the highly specialized echolocation system. Finally, I discuss where bats fit into our current models—birds, anurans, cetaceans, and rodents—and how using a comparative approach can greatly expand our understanding of acoustic communication.

11:40–12:00 Panel Discussion

FRIDAY MORNING, 6 NOVEMBER 2015 CITY TERRACE 7, 9:00 A.M. TO 10:30 A.M.

Session 5aAO

Acoustical Oceanography and Underwater Acoustics: Topics in Acoustical Oceanography

Gopu R. Potty, Chair
Ocean Engineering, University of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882

Contributed Papers

9:00

5aAO1. Analytical discussion of past measurements of acoustic attenuation in mud sediments and of possible future experimental approaches. Allan D. Pierce (PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Attenuation of compressional waves in mud is higher than in sea water, and less than in sandy/silty sediments. For experiments reported in 1964 (Acustica) by Wood and Weston, the Inferred circumstances are a 1 m thick mud slab overlying a gravel bottom, with air above (low tide), and with source and receiver either at the mud-gravel interface or slightly above it. Frequencies from 4 kHz to 72 kHz were transmitted, with signals received at a succession of horizontal distances of up to 50 m. Reported data in decibels versus range show considerable erratic behavior. Attempt is made to explain this behavior using a full-wave analysis of a Pekeris waveguide, with a fluid layer (the mud slab) overlying a (fluid or elastic) half-space, with a series of plausible guesses concerning the properties of the half-space. The conclusion is that the data and the circumstances allowed considerable variability in the attenuation estimates, but those made by Wood and Weston were about as good as could be expected. it is argued that placing source and receiver near the interface was a relatively poor choice, and that future experiments of this general type might give better results with the use of vertical arrays.

9:15

5aAO2. Experimental design for sediment characterization in the New England Mud Patch. Gopu R. Potty, James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), and James F. Lynch (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Wood Hole, MA)

This study focuses on the design of an experiment to estimate the shear wave properties of ocean bottom sediments at a location in the southern

5a FRI. AM

New England Continental Shelf called the "New England Mud Patch." The mud patch is a 13,000 square kilometer area covered by fine-grained sediment. The inversion technique is based on collecting interface wave (Scholte wave) data using geophones on the sea bottom. The data for the inversion consist of Scholte wave phase velocity dispersion calculated from the geophone array data. The present study aims at applying this Scholte wave based shear wave inversion technique to the "mud patch" area. Two different interface wave measurement systems will be presented. Appropriate source to receiver ranges will be explored based on simulations. The engineering challenges associated with deploying the system in a soft seabed will be investigated and design modifications will be investigated. Acoustic and sediment data from the 1996 Shelf Break Primer experiment, which was conducted in the western side of the mud patch, will be reviewed. The simulations will be based on historic sediment and acoustic data. [Work sponsored by Office of Naval Research, code 322 OA.]

9:30

5aAO3. Comparisons between a spherical aggregation of scatterers, and hard and soft spheres using single-frequency and pulsed signals. Adaleena Mookerjee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., 2010 Autolab, Ann Arbor, MI 48109, adaleena@umich.edu)

When sound is projected into the ocean, the backscattered signal may provide information about the object(s) from which the sound has scattered. When the backscattered sound comes from an aggregation of strong scatterers, such as a school of fish at their swim-bladder resonance frequency, a phenomenon called Coherent Backscatter Enhancement (CBE) may occur, and this phenomenon could aid in discriminating fish schools from other similar-strength scatterers in the ocean water column. When CBE occurs, the addition of in-phase scattered waves from propagation path pairs produces a scattered field intensity enhancement of as much as two in the direction opposite to that of the incident wave. This presentation describes the results of CBE simulations of spherical aggregations of scatterers based on the Foldy (1945) equations, and provides comparisons to backscattering from single perfectly reflecting spheres. Interestingly, a spherical aggregation of 4200 strong scatterers with wave number scaled radius $ka = 32$ may provide backscattering equivalent to that from a single larger perfectly-reflecting sphere with $ka = 53$. Additional simulation comparisons involving the statistics and time histories of harmonic and frequency sweep incident waves are shown. [Work supported by the Office of Naval Research and by Advanced Research Computing at the University of Michigan.]

9:45

5aAO4. Compressive ocean acoustic sound speed profile estimation in shallow water. Michael Bianco and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbianco@ucsd.edu)

The estimation of ocean acoustic sound speed profiles (SSPs) requires the inversion of an acoustic transmission model using limited observations. Provided the parameters of the inverse model are sparse, Compressive Sensing (CS) can help solve such underdetermined problems accurately, efficiently, and with enhanced resolution. Here, CS is used to estimate range-independent acoustic SSPs in shallow water ocean environments using a normal-mode representation of the acoustic field. Two sparse parameterizations of the SSPs are considered. The first parameterization assumes that change in the sound speed with depth is sparse and that the SSP can be constructed from a limited number of dominant changes in sound speed. The second case assumes *a priori* information about the ocean SSP variability in terms of Empirical Orthogonal Functions (EOFs), estimating the dominant EOFs describing the current SSP. For both cases, both real and synthetic acoustic data are processed. It is shown that in the CS framework, both of the optimizations can be solved with increased resolution and robustness over traditional methods.

10:00

5aAO5. Using broadband acoustic signals and modal behavior to assess scales of ocean variability in shallow water waveguides. Mohsen Badiey (College of Earth, Ocean and Environment, The University of Delaware, 107 Robinson Hall, Newark, DE 19716, badiey@udel.edu) and Marshall H. Orr (College of Earth, Ocean and Environment, The University of Delaware, Bryantown, MD)

The sound speed fields of the continental shelf water column are anisotropic and have "fronts" related to propagating internal wave packets. The angle between a broadband acoustic signal's source–receiver propagation track and the propagating "fronts" affects its modal composition. If the temporal variability of the modal behavior of the acoustic signals are monitored in time and space, one might identify the fluid process that causes the changes to the sound speed field. In this paper, we first present a heuristic view of how the modal properties of broadband signals can be affected by the sound speed field variability caused by propagating internal wave packets. We then present three days of acoustic field data acquired on vertical and horizontal arrays placed in ~100 m of water about 20 km from the location of acoustic signal sources. Temporal changes in the received acoustic signal's modal composition will be compared to numerical simulations. The changes in modal composition will be correlated to internal wave induced variability in the horizontal refraction of the signals.

10:15

5aAO6. Measurements and modeling of the variability in normally incident reflections from the seabed. Saroj Bardewa and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR 97201, saroj@pdx.edu)

Normal incidence reflection measurements taken during the TREX13 experiment off the coast of Florida show substantial variability. One likely source of variability is due to scattering from the rough surface of the seabed. These data were taken from an omnidirectional source at a relatively low frequency (around 3 kHz). Systems being considered for measuring seabed properties (such as sound speed, density, and roughness) may use commercial echo-sounders that are directional and operate at much higher frequencies. Measurements from the TREX13 data as well as higher frequency systems are analyzed together with modeling to determine how surface roughness effects these measurements. The modeling uses an integral equation approach with a power law roughness spectrum to calculate backscattering of incident plane waves. The effects of changing the model parameters such as seabed type and roughness on the nature of backscattering are examined. The simulated results are compared with the measurements obtained from the experiments. The measurements are further used to calculate the statistical parameters to understand the nature of the normal reflection from a rough seabed.

Session 5aMU

Musical Acoustics: General Topics in Musical Acoustics

Martin S. Lawless, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

James P. Cottingham, Cochair
Physics, Coe College, 1220 First Avenue, Cedar Rapids, IA 52402

Contributed Papers

8:30

5aMU1. The inner ear as a musical instrument. Brian Connolly (Music Dept., Maynooth Univ., Logic House, Maynooth Co. Kildare, Ireland, bconnolly1987@gmail.com)

This paper addresses the place of the ears and their potential as musical instruments. Psychoacoustics research into the non-linearities of the inner ear has proven that the organ has much more to offer the composer than has been previously considered. By reversing the role of the ear from being a submissive receiver to an active participant in the creative process, an exciting level of opportunity opens up for the composer. A focus is given in this paper to bandwidth phenomena and auditory distortion products with references to the author's own compositional material which seeks to highlight that the investigation of the non-linear nature of the inner ear is revolutionary in relation to the place of the ear within contemporary composition. Central to this work is the notion that the ear is not simply a passive component of the musical experience. Adorno once stated that the ear is a "dozy and inert" organ and by disproving such a view one can destabilize common assumptions about psychoacoustics and the potential of the ears as musical instruments.

8:45

5aMU2. Input impedance of sheng pipes augmented with resonators. Matthew F. LeDuc (Phys. Dept., Univ. of Michigan, Ann Arbor, MI 48109, mafrledu@umich.edu), Nathan K. Haerr, Chevelle N. Boomershine, and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

The sheng is a free reed mouth organ constructed of bamboo pipes with a free reed in each pipe near one end. A version commonly available today has 17 pipes in a circular arrangement with the reeds enclosed in a wind chamber. The pipes are nearly cylindrical over a significant portion of their length, but the bore near the reed end becomes approximately conical. A traditional instrument of China, the sheng, has a long recorded history, but a number of enlarged and modernized versions were created in the twentieth century, some featuring keys and additional pipes. The sheng in the current study is a standard 17-pipe configuration, with cylindrical metal pipe resonators attached to most of the pipes. These resonators amplify the radiated sound and also alter the tone quality. Calculations of input impedance have been made for the pipes, with and without the resonators attached, taking into account the position of the reed along the pipe, tuning slots, finger holes, and non-circular pipe cross sections. These calculations are compared with the measured pipe impedances as well as the measured sounding frequencies and sound spectra. [Research supported by NSF REU grant PHY-1004860.]

9:00

5aMU3. Empirical study of violin acoustics and its perception under various mutes. Sina Mousavion (SciTec, Ernst Abbe Fachhochschule-Jena, Jakobsplan 1, Nr. 02-09-07-0, Weimar 99423, Germany, sina.mousavion@yahoo.com) and Suchetana Sarkar (SciTec, Ernst Abbe Fachhochschule-Jena, Heidelberg, Germany)

Violinists often practice with mutes, the purpose of which is to reduce the volume so the sound does not disturb other people. There is however a great disadvantage; practice mutes not only lower the volume but change the timbre dramatically, making it unpleasant to the ear. Previous researches mostly focus on how to find the elusive "Stradivarius Sound." On the other hand, via sound analysis, we empirically study different mutes, in order to identify the markers that allow the violin to retain its "Signature Sound." The general problem lies in the complexity of violin sound propagation and its perception. Here, we show that all types of mutes necessarily diminish the frequency band of 1.5–2.5 kHz, which tips the balance of the sound toward a darker color. The results demonstrate that unless a proportional balance between the amplitude of the harmonics of each tone is maintained, the sound loses its Brilliance character. In addition, some mutes tend to add spurious resonance peaks at lower frequencies, which further emphasizes the imbalance of tones in the lower regions. These factors can aid in the development of a mute that does not inhibit the intrinsic characteristic of the violin.

9:15

5aMU4. Acoustic and structural resonance characteristics of the cajon. John Pehmoeller and Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 48504, pehm3247@kettering.edu)

The cajon is a percussion instrument with origins in 16th century Peru. Literally translated "crate," the cajon was originally a crate that players sat upon and struck to make rhythmic patterns. The modern version is a wooden box, closed except for a hole in the back panel. The front panel is struck with the hands or one of several types of beaters. The front panel is thinner than the others and is often attached with a slightly loose top edge for a snare-like effect. Alternatively, the snare sound is obtained with wires or guitar strings placed against the inside surface of the front panel, and these rattle when the panel is struck. Measurements of acoustic transfer functions with impulse excitation and roving hammer structural measurements explore the acoustic and structural resonances of the cajon with rattling strings removed. Spectra from recordings establish timbral features of the instrument, both with and without the rattling strings. By comparing results from measurements and recordings, the important features of the particular instrument tested in this study are identified, and possible directions for improved the designs are suggested.

9:30

5aMU5. Improving reliability of measurement algorithms in differentiated electroglottograph and audio signals. Shonda Bernadin (Elec. and Computer Eng., Florida A&M University-Florida State Univ. College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, bernadin@eng.fsu.edu) and Richard Morris (Speech and Commun. Disord., Florida State Univ., Tallahassee, FL)

The purpose of this work was to develop an optimized algorithm for determining the closed quotient (CQ) measurements of the vocal folds of female singers. This study extends previous results which showed that the measurement algorithm for differentiated electroglottograph (dEGG) signals and differentiated audio (dAUDIO) signals gave moderately reliable results with a correlation coefficient of $r = 0.79$. Additional performance studies indicate that the reliability may be improved by combining the differential signals and modifying the measurement algorithm to reduce the amount of signal overlap. Preliminary results using the modified measurement algorithm yield a significant increase in the correlation coefficient ($r = 0.83$). This works contributes to the development of a computational framework for determining accurate and reliable measurements for characterizing register shifts through an octave in female singers using electroglottography.

9:45

5aMU6. Impact of acoustic resonances on overtone correlations across a large musical instrumental database. Sarah R. Smith and Mark F. Bocko (Univ. of Rochester, 405 Comput. Studies Bldg., P.O. Box 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

When performers use vibrato, the pitch and intensity of the note is modulated in order to create an expressive effect. Although one might expect that vibrato would produce a proportional frequency modulation in each of the overtones, this is not always the case. Past work has shown that the instantaneous frequencies of some instruments, particularly bowed strings, tend to exhibit reduced overtone correlation when performed with vibrato. Additionally, the presence of reverberation in a recording has been shown to reduce the correlation between overtones for many instruments. In this study, we present an analysis of overtone correlations for a wide range of instrument sounds taken from multiple large databases. Using this large library of tones, we illustrate the effect of instrument resonators on these correlations for tones performed both with and without vibrato.

10:00–10:15 Break

10:15

5aMU7. Complex point source model to calculate the sound field radiated from musical instruments. Tim Ziemer and Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, tim.ziemer@uni-hamburg.de)

A simple method is described to record the radiated sound of musical instruments and to extrapolate the sound field to distances further away from the source. This is achieved by considering instruments as complex point sources. It is demonstrated that this simplification method yields plausible results not only for small instruments like the shakuhachi but also for larger instruments such as the double bass: The amplitude decays in a given manner and calculated interaural signal differences reaching the listener decrease with increasing distance to the source. The method is applied to analyze the sound radiation characteristics as well as the radiated sound field in a listening region regardless of room acoustical influences. Furthermore, it has been implemented in a psychoacoustic wave field synthesis system to generate the impression of a certain source width. Implementations in terms of room acoustical simulations, spatial additive synthesis and sound field synthesis are discussed.

10:30

5aMU8. Stroboscopic illumination for electronic speckle pattern interferometry. Colin Gavin and Steve Tufte (Dept. of Phys., Lewis & Clark College, MSC 15, 0615 SW Palatine Hill Rd., Portland, OR 97219, tufte@lclark.edu)

A common realization of holographic interferometry is called Electronic Speckle Pattern Interferometry (or ESPI)—a technique capable of measuring harmonic modes of vibrating objects. We present a method of improving the contrast and quality of fringe patterns recorded with a simple, table-top ESPI system. In particular, by using stroboscopic illumination generated by an optical chopper, we are able to produce fringes that follow a cosine pattern, rather than the Bessel pattern fringes that result from time averaging. Since Bessel function amplitudes rapidly decrease for subsequent maxima, the stroboscopic cosine fringes show much better contrast. Also, because the zeros of cosine fringes are evenly spaced, it is much simpler to interpret the images to extract quantitative deformation amplitudes. We show that our results agree well with the theoretical predictions. This system was developed for use in musical acoustics research as a Senior Thesis project by an undergraduate student (Gavin). This low-cost, simple modification of the commonly used ESPI system could benefit other colleges and universities using holographic interferometry for acoustics research.

10:45

5aMU9. An experimental modal analysis on the coimbra model of the Portuguese guitar. Octávio Inácio and Rui Ribeiro (ESMAE, Rua da Alegria, 503, Porto 4000-053, Portugal, octavioinacio@esmae.ipp.pt)

The most distinctive Portuguese traditional music style is Fado. In this form of Portuguese music, a singer is accompanied by two instruments: a classical guitar and a pear shaped plucked chordophone with six courses of double strings—the Portuguese guitar. There are two distinct types of this instrument—the Lisbon and the Coimbra models—named after the towns where the two different styles of Fado have developed. These guitars differ basically on their size and tuning, both comprising 6 orders of double steel strings, while the construction method (strutting patterns, wood species used, and soundboard thickness distribution) vary for different builders. As part of an ongoing research project that investigates the vibroacoustical behavior of this instrument for different types of designs, an experimental modal analysis of a fully assembled Coimbra guitar was performed. In this work, we present the results of this analysis showing the main characteristics of the frequency response curves and significant vibratory modes as compared to other similar plucked-string instruments.

11:00

5aMU10. Timbre aspects of ride cymbals: Sound coloration analysis using psychoacoustics models and subjective evaluation. Kauê Werner, Erasmo F. Vergara, Stephan Paul, and Júlio A. Cordioli (Lab. of Acoust. and Vib., UFSC, Acari Silva 74, Casa 3, Florianopolis, SC 88035440, Brazil, kaue.werner@lva.ufsc.br)

The timbre of a musical instrument is a multidimensional variable that relies on the complexity of human perception. So far, there are only a few studies concerning the timbre of percussion instruments. The sound characteristics of cymbals depend on geometry, material, ways of playing, and so on. On the perceptive side, sound coloration is one of the dimensions most used by musicians with regard to compare the timbre of Ride Cymbals. In order to get insight into the perception of the sound of Ride Cymbals, subjective evaluations of recorded cymbal sounds were carried out, using bipolar rating scales. Psychoacoustic models, such as Sharpness and Tonality, were also used to analyze the sound signals. Based on subjective and objective evaluations, linear regression models were proposed to quantify perceptive dimensions such as brightness and darkness. The coefficients of determination indicate that this kind of model can be used as a feasible way to represent sound coloration in Ride Cymbals.

11:15

5aMU11. Construction of a Finite Element Model of the Japanese *koto* and its comparison with the reference instrument. A. Kimi Coaldrake (Music, Univ. of Adelaide, Elder Conservatorium of Music, Adelaide, SA 5005, Australia, kimi.coaldrake@adelaide.edu.au)

As a first step toward developing a generic model of the Japanese *koto* (13 string plucked zither) and related Asian zithers to understand the physical characteristics of the instruments and how they contribute to the overall sound quality, a finite element model of the *koto* was constructed. Samples of paulownia wood from a reference instrument were examined by SEM to establish the physical properties of this less well-characterized wood. The model was validated against the chladni patterns of Ando's 1986 experiments. Eigenfrequencies and mode shapes were calculated before coupling the instrument to a sphere of air. Frequencies corresponding to the standard tuning of the 13 strings were applied at the same point where plucking normally occurs. A steady state response of the instrument was recorded that showed standing waves within the cavity of the instrument. Frequency scans were also undertaken. Finally, the transient response of the model to the string frequencies were measured and compared with the playing of the reference instrument. Overall, the model predicts the main features and some subsidiary ones of the reference instrument and point to directions for future work for characterizing the sound quality of the *koto*.

11:30

5aMU12. Timbre maps to characterize the sound quality of the Japanese *koto*. A. Kimi Coaldrake (Music, Univ. of Adelaide, Elder Conservatorium of Music, Adelaide, SA 5005, Australia, kimi.coaldrake@adelaide.edu.au)

The Japanese *koto* and related zithers in East Asia are known for their distinctive sound quality (timbre), but the precise origins of the characteristics remain elusive. In an attempt to extend the knowledge of these characteristics, a finite element model of the Japanese *koto*(13 string plucked zither) was constructed using Comsol Multiphysics 5.1 based on a *koto* of known provenance and parameters. A series of experiments to identify the acoustical properties of the instrument were compared against results of the model. The results were then collated into a series of timbre maps to establish patterns of responses. This included overlaying eigenfrequencies, individual peaks from the spectra of the thirteen strings for the standard (*hirajō shi*) tuning, hidden peaks identified by the fourth derivative frequency scans and spectra from mechanical tapping of the reference instrument. The study shows that the anisotropy of the paulownia wood of the *koto* plays an important role in the vibration modes and frequency response spectrum. The timbre maps then identifies characteristics of the non harmonic responses that contribute to the *koto*'s distinctive sound and provide a template for the study of sound quality associated with culturally specific timbral preferences for East Asian zithers.

FRIDAY MORNING, 6 NOVEMBER 2015 GRAND BALLROOM 1, 9:00 A.M. TO 10:00 A.M.

Session 5aNS

Noise: Noise Potpourri

Scott D. Sommerfeldt, Chair
Dept. of Physics, Brigham Young University, N181 ESC, Brigham Young University, Provo, UT 84602

Contributed Papers

9:00

5aNS1. Development of open air sound barrier system using single microphone and speakers. Natsuki Takao (School of Sci. for Open and Environment Systems, Keio Univ. Graduate School of Sci. and Technol., Hiyoshi 3-14-1, Kohoku-ku, Yokohama 223-8522, Japan, n.takao@z6.keio.jp)

Music players usually build soundproof rooms in their houses. These rooms prevent noise problems among their neighbors when they practice at home. However, in soundproof rooms, vibrancy of musical sound is often lost, and music players feel as if the sound is cooped-up. Besides, installation cost of a soundproof room is usually high. We therefore aim to create a new sound barrier system. This system produces soundproof space in the open air by using single microphone and adequate number of speakers. The speakers cancel propagating waves which come from the musical instruments by outputting antiphase waves. The sound is muffled in the half sphere surrounding the player. Thus, the player can listen to their sound without losing vibrancy, while sound leakage outside the system is suppressed. Antiphase waves are predicted based on the wave equation with the signal picked up by the microphone as the boundary condition. In this presentation, a simulation for determination of the adequate number and position of the speakers is presented. Then, experimental validation is discussed in detail.

9:15

5aNS2. Studying of the noise sources in a pneumatic nail-gun process. Zahra Nili Ahmadabadi, Frédéric Laville, and Raynald Guilbault (Mech. Eng., Université du Québec, École de technologie supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, zahra.nili-ahmadabadi.1@ens.etsmtl.ca)

Despite generating high noise levels responsible for hearing loss among workers, nail-guns have been used to connect wood pieces since the 50s. The present study belongs to a broader investigation aiming to reduce noise emissions in nail-guns. This noise reduction objective may be achieved by a nail-gun concept design improvement. This requires a study of the noise sources in time domain. The study uses an advanced measurement setup to identify the existing noise sources and their causes in each time interval. The setup includes nine microphones, two accelerometers, two pressure transducers, and a high-speed camera. Three major noise sources were identified during the nailgun process: the air exhaust, the body of the machine, and the workpiece. The air exhaust noise is radiated from the air exhaust holes mostly before the nail driving operation and during the air exhaust process. The noise radiated from the body of the machine is caused by vibrations of different internal/external parts of the machine and air movements. It persists almost throughout the whole duration of the nailgun process. Finally, the workpiece noise is radiated from the vibrating workpiece starting simultaneously with the nail driving operation and ending before the start of the air exhaust process.

9:30

5aNS3. Improved ear simulator for extended application range. Per Rasmussen (G.R.A.S. Sound & Vib. A/S, Skovlytoften 33, Holte 2840, Denmark, pr@gras.dk), Jacob Soendergaard (G.R.A.S. Sound & Vib. NA, Twinsburg, OH), and Morten Wille (G.R.A.S. Sound & Vib. A/S, Holte, Denmark)

The international standard IEC 60318-4 specifies an occluded ear simulator for testing headphones, earphones, hearing protectors, hearing aids etc. The standard specifies a specific microphone type which limits the dynamic range of the coupler, such that it is not possible to measure very low levels or very high levels. Additionally, the standard 711 coupler is often interfaced to a pinna simulator incorporated in a Head and Torso simulator as per IEC 60318-7. This interface has traditionally been implemented as a cylindrical, straight ear canal simulator. This makes the fit of many modern in-ear headphones and hearing protectors problematic and unrealistic. By using low noise microphones instead of the standard microphones, the coupler can be used for measuring extremely low sound pressure levels such as noise floor, low level distortion or microphonics. Conversely, by using low sensitivity microphones, the coupler can be used for extremely high level measurements—useful for testing active and passive attenuation ratings of hearing protectors. Moreover, using a vast database of 3D human ear canal scans, a new pinna and ear canal simulator is proposed that will greatly improve measurement accuracy and repeatability on products going on or in the ear.

9:45

5aNS4. Modeling of acoustic resonators and resonator arrays with non-ideal geometries. Matthew F. Calton and Scott D. Sommerfeldt (Brigham Young Univ., N306 ESC, Provo, UT 84602, mattcalton@gmail.com)

Acoustic resonators offer stable, cost-effective attenuation for many noise control applications. Due to their widespread use in engineering and physics, analytical expressions with varying degrees of accuracy have been obtained to predict performance. However, most of the existing formulations are limited to ideal geometries with sharp transitions between two regions. Practical applications of these acoustic resonators often involve space limitations that necessitate curved segments and gradual transitions. This research aims to better characterize the response of non-ideal geometry acoustic resonators. Using currently available expressions for duct bends, the acoustic impedance of bends with various angles and lengths is determined, in an effort to develop relatively simple models for incorporating such elements in a resonator. Other non-ideal geometries as they relate to Helmholtz resonators and arrays of resonators are also explored. These calculations are compared to experimental impedance measurements for validation of the model.

FRIDAY MORNING, 6 NOVEMBER 2015 — CLEARWATER, 8:00 A.M. TO 11:45 A.M.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics II

Michael R. Haberman, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

8:00

5aPA1. Study of Wigner crystal in n-GaAs/AlGaAs by surface acoustic waves. Alexey V. Suslov (NHMFL, Florida State Univ., 1800 E. Paul Dirac Dr., Tallahassee, FL 32310, suslov@magnet.fsu.edu), Irina L. Drichko, Ivan Y. Smirnov (A. F. Ioffe PTI of RAS, St.-Petersburg, Russian Federation), Loren N. Pfeiffer (Elec. Eng., Princeton Univ., Princeton, NJ), Ken W. West (PRISM, Princeton Univ., Princeton, NJ), and Yuri M. Galperin (Depertment of Phys., Univ. of Oslo, Oslo, Norway)

The surface acoustic wave (SAW) technique was used for studying a 65 nm wide GaAs quantum well containing a two dimensional electron system (2DES) with the density of $5 \cdot 10^{10}$ cm^{-2} and the mobility of $8 \cdot 10^{6}$ cm^2/V·s. A SAW propagated in proximity of the quantum well. Both SAW attenuation α and SAW velocity V were measured in perpendicular magnetic field of up to 18 T in the frequency f range 28.5–306 MHz at temperatures 40–380 mK. Then, both real and imaginary parts of the ac conductance $\sigma(f) = \sigma_1(f) + i \cdot \sigma_2(f)$ of the 2DES were calculated from α and V. Dependencies of α, V, σ_1, and σ_2 on magnetic fields showed a rich oscillation pattern, in particular, the fractional quantum Hall effect (FQHE). Thus, at low temperature, the FQHE state was observed at the filling factor $\nu = 0.20$, whereas, in vicinity of this value, at $\nu = 0.19$ and 0.21, the 2DES behaved as a 2D Wigner crystal weakly pinned by a disorder random potential. The calculated pinning frequencies at high and zero magnetic fields were 193 MHz and 12.4 GHz, respectively, and the correlation length of the random potential was 10^{-2} cm. [Work supported by RFBR 14-02-00232, NSF DMR-1157490, the State of Florida, the Gordon and Betty Moore Foundation GBMF2719, and NSF MRSEC-DMR-0819860.]

8:15

5aPA2. The acoustic study of soil liquefaction effects in-situ. Andrey Konkov, Andrey Lebedev, and Sergey Manakov (Geophysical Acoust., Inst. of Appl. Phys., 46 Ul'yanov St., Nizhny Novgorod 603950, Russian Federation, magister44@yandex.ru)

This study is aimed to describe effects of saturation with water on soil stiffness and bonds strength. Linear and nonlinear soil acoustic characteristics as dependent on saturation have been studied in-situ. At the end of experiment, the soil became unstable that allowed tracking the saturation process as a whole. The experiment was prepared as follows. Two weeks before the experiment fulfilment, the trench was dug and a hose with holes was embedded. The hose was connected to the water supply network through the water meter. During saturation, the soil acoustic responses were being continuously recorded. The soil linear stiffness has been measured by remote technique based on the analysis of the dispersion of Rayleigh wave polarization. The measurements were carried out by pairs of vertically and horizontally oriented geophones. Then, the bulk stiffness and the Poisson's ratio profiles were reconstructed. Special attention was given to the assessment of its nonlinear parameters. Effects of hysteresis nonlinearity in dependence "force versus displacement" were observed and discussed as well. The force and displacement were measured by two accelerometers mounted on the baseplate and the reaction mass of vibrational source. [The work was partially supported by RFBR, research projects No.14−02−00695, 15−05−08196, 15−45−02450, and 14−05−31249.]

8:30

5aPA3. Scattering by a thin and flat elliptic object: How the conformal mapping approach allows addressing numerical issues. Didier Cassereau (Laboratoire d'Imagerie Biomédicale, 15 rue de l'Ecole de Médecine, Paris 75006, France, didier.cassereau@upmc.fr), Fabien Mézière, Marie Muller, Emmanuel Bossy, and Arnaud Derode (Institut Langevin, Paris, France)

In this work, we are interested in modeling the propagation of an ultrasonic field in the complex trabecular bone structure. For this purpose, we use a simplified formulation, based on a representation of the bone tissue as thin and flat elliptic scatterers. With an ultrasonic inspection frequency of 1 MHz, the largest dimension of the ellipse is about half a wavelength, while the smallest dimension is 7–10 times smaller. The computation of the scattered field is based on a modal decomposition, generalized to the case of scatterers of arbitrary geometry (Chati *et al.*, "Modal theory applied to the acoustic scattering by elastic cylinders of arbitrary cross section," J. Acoust. Soc. Am. **116**, 2004). Due to the very thin and flat geometry, this approach suffers huge numerical difficulties that require the use of enhanced precision and very long computation times. In this paper, we illustrate these issues. An alternative has been proposed (Liu *et al.*, "Conformal mapping for the Helmholtz equation: Acoustic wave scattering by a two dimensional inclusion with irregular shape in an ideal fluid," J. Acoust. Soc. Am. **131**, 2012), based on a conformal mapping of the ellipse. We compare the two formulations and show how the conformal mapping allows reducing drastically the numerical issues resulting from the standard approach.

8:45

5aPA4. Parametric study of an ultrasonic non-destructive testing problem based on the reciprocity principle and a hybrid numerical method. Florian Lyonnet, Didier Cassereau (Laboratoire d'Imagerie Biomédicale, Laboratoire d'Imagerie Biomédicale, Paris, France, florian.lyonnet@upmc.fr), and Marie-Françoise Cugnet (AREVA NP, Ugine, France)

Numerical simulations are needed for both the design and the improvement of Ultrasound Non-Destructive Testing techniques in industry. One typical configuration is the pulse echo immersion testing of a material with a possible crack. This is a large and multiscale elastodynamic problem, for which the computation may be intensive using standard discrete methods. Different approaches have been proposed to gain efficiency in terms of the computation time, as i) the hybridization of a discrete method with a semi-analytical formulation or ii) using the reciprocity principle to separate the calculation of the field radiated by the crack from the field in the same material. We present a numerical strategy based on those two approaches in order to perform 3D simulations with a possible parametrization of the crack geometry and location. First, the Spatial Impulse Response (SIR) of a focalized transducer in water is computed using a semi-analytical software. Then, we show that the calculation of the SIR can be pursued in the same elastic material using a FDTD method. To take into account the effects of the crack, this SIR is used in conjunction with the time domain reciprocity equation. This process allows performing an efficient computation of the echo signal from the crack. Results and comparisons with experimental measurements will be presented in a 2D case.

9:00

5aPA5. Calculation of vibrational mode contributions to sound absorption in excitable gases by decomposing multi-relaxation absorption curve. Ke-Sheng Zhang (School of Information Eng., Guizhou Inst. of Technol., 1st, Caiguan Rd., Yunyan District, Guiyang, Guizhou 550003, China, keshengzhang@163.com), Ming Zhu (School of Electron. Information and Communications, Huazhong Univ. of Sci. and Technol., Wuhan, China), Chun Li (Library, Guizhou Inst. of Technol., Guiyang, China), and Weihua Ou (School of Mathematics and Comput. Sci., Guizhou Normal Univ., Guiyang, China)

Molecular vibrational relaxation is responsible for the sound relaxational absorption in most excitable gases. However, it is desirable to calculate the contribution of each vibrational mode to sound multi-relaxation absorption. In this paper, first, a sound multi-relaxation absorption curve is decomposed into the sum of single-relaxation curves; second, based on this decomposed characteristic, a model to quantitatively analyze the vibrational mode contributions to sound absorption is proposed. The simulation results quantitatively demonstrates that the primary relaxation process connected with the lowest mode is the decisive factor for sound relaxational absorption, and the mode with lower vibrational frequency supplies higher contribution to the primary relaxation process in pure polyatomic gases.

9:15

5aPA6. Measurement of tortuosity and viscous characteristic length of double-layered porous absorbing materials with rigid frames via transmitted ultrasonic wave. Mustapha Sadouki (Département de Sci. de la Matière, Université Djilali Bounaama à khemis-miliana, Rte. Thenia el Had, Ain Defla, Khemis-miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz), Amine Berbiche (Fac. de Physique, USTHB, Algiers, Bab Ezzouar, Algeria), Mohamed fellah (Fac. de Physique, USTHB, Alger, Algeria), Zine El Abidine fellah (LMA UPR7051 CNRS Aix-Marseille Univ, Centrale Marseille, Marseille, France), and Claude Depollier (Laboratoire d'Acoustique de l'Université du Maine UFR STS, LUNAM Université du Maine, UMR CNRS 6613, Le Mans, France)

In this work, an indirect method is proposed for measuring simultaneously acoustic parameters describing the ultrasonic propagation in double-layered porous medium. The porous media consist of two slabs of homogeneous isotropic porous materials with a rigid frame. Each porous slab is described by equivalent fluid model, in which the acoustic wave propagates only in the fluid saturating the material. The inverse problem is solved numerically using experimental transmitted waves in time domain. Four parameters are inverted: tortuosity and viscous characteristic lengths of the two layers. Tests are performed using industrial plastic foams. Experimental and numerical validation results of this method are presented.

9:30

5aPA7. Acoustic transmission from a single crystal layer of nonlinear resonators in a soft elastic matrix. Nicolas Viard and Nicholas Fang (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, nviard@mit.edu)

Ultrasound transmission measurements of a single crystal layer of identical nonlinear resonators are reported for frequencies under 1 MHz. The resonators are spherical gas-filled cavities embedded in a soft elastic matrix. The cavities are formed by injection of gas through a capillary tube in a polymer solution right before it cures. Sample characteristics such as the size or the spacing between the cavities are adjusted by varying the injection parameters. Typical size for the cavities ranges from 100 μm to 1 mm. The sample is immersed in water where transmission measurements are conducted. As allowed by acoustic transducer, amplitude and phase of the wave is measured in transmission as a function of the frequency and the amplitude of the initial incident pulse.

9:45–10:00 Break

10:00

5aPA8. Stress and energy transmission by inhomogeneous plane waves into dissipative media. Daniel C. Woods, J. Stuart Bolton, and Jeffrey F. Rhoads (School of Mech. Eng., Purdue Univ., 585 Purdue Mall, West Lafayette, IN 47907, woods41@purdue.edu)

The characteristics of sound transmission into real, or dissipative, media differ from those of transmission into lossless media. In particular, when a plane wave in a fluid is incident upon a real, dissipative elastic material, the transmitted waves are in general inhomogeneous, even when the incident wave is itself homogeneous and incident at a sub-critical angle; and more significantly, energy transmission occurs even above the critical angle. In addition, for any real incidence angle, the parameters of an incident inhomogeneous wave may be tuned so that there is no reflection from the surface of a viscoelastic solid. That phenomenon may be exploited in applications requiring energy transmission into solids. In this work, the transmission of incident inhomogeneous, as well as homogeneous, acoustic waves into solid materials is characterized; a hysteretic damping model is assumed. Numerical results are presented for the transmitted stress and energy distributions

for typical solid materials, including polymer-based solids. The conditions for total transmission, i.e., no reflection at the interface, are explored, where the propagation angle, degree of inhomogeneity, and frequency of the incident wave are varied for a given material. These investigations show substantial transmission gains in the vicinity of the zero of the reflection coefficient, compared to homogeneous incident waves.

10:15

5aPA9. Standoff photoacoustic spectroscopy for hazard detection. Logan S. Marcus, Ellen L. Holthoff, and Paul M. Pellegrino (Sensors and Electron Devices Directorate, U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-SEE-E, Adelphi, MD 20783, loganmarcus@gmail.com)

Photoacoustic spectroscopy (PAS) is a versatile and sensitive chemical sensing method. This versatility allows for the construction of a variety of sensors that are optimized for specific sensing tasks. Current research at the U.S. Army Research Laboratory (ARL) is focused on the development of a standoff hazardous materials detection technique based on an interferometric sensor. The standoff detection paradigm increases operator safety and reduces sample preparation requirements as compared to traditional photoacoustic cell-based sensors. We demonstrate the collection of photoacoustic spectra of layered solid samples at a standoff distance of one meter. The layered samples are constructed via deposition of a thin layer of energetic or other hazardous substance upon a thick substrate. We will also discuss excitation source selection as it relates to the operating mode of the source (i.e., pulsed or continuous wave (CW) modulated).

10:30

5aPA10. A self consistent theory for phonon propagation in a suspension of one-dimensional filaments. Douglas Photiadis (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, douglas.photiadis@nrl.navy.mil)

Using a self consistent, field theoretic multiple scattering theory in a viscous, acoustic fluid, we have developed a mean field model to describe the propagation of phonons in suspensions of one dimensional filaments. This geometry is approximately realized by the cytoskeleton in generic cells and can be studied in a controlled manner using suspensions of carbon nanotubes. We have extended the coherent potential approximation method, typically used for point scatterers, to the case of one dimensional filaments using the supersymmetric method, an approach employed with great success in disordered electronic systems. Unlike similar systems involving suspensions of effectively point scatterers, viscosity is found to play an important role in determining the observed wave speed because of both the high density and the one dimensional nature of the scatterers. We have carried out Brillouin light scattering measurements on both live cells, for which we believe the observed frequency shifts result from coupling to the cytoskeleton, and carbon nanotube suspensions. By varying the temperature, we may approach the glass transition of the fluid and substantially change the viscosity to directly test our predictions. Preliminary measurements show frequency shifts in accord with the predictions. [Research funded by the Office of Naval Research.]

10:45

5aPA11. Cost-effective potential application of acousto-optic Bragg imaging of biological tissue. Alem Teklu (Phys. & Astronomy, College of Charleston, 9 Liberty St., Charleston, SC 29424, teklua@cofc.edu), Nico Declercq (Georgia Inst. of Technol., Metz, France), and Michael McPherson (Div. of Natural Sci., Northwest MS Community College, Senatobia, MS)

Acousto-Optic Bragg Imaging is a technique that uses the interaction of light with ultrasound to optically image obstructions in acoustical fields. Existing reports of Acousto-Optic Bragg Imaging based on transmission of acoustic fields through obstructions exhibit strong acoustic impedance mismatches manifested by poor image quality and missing details of physical structures of obstructions. In this work, the image quality was improved to exhibit detailed physical structures of an object by using an improved Bragg imaging system. This project investigates the possibility of extending an acoustic Bragg imaging technique in transmission mode to image animal or plant tissues; a small Azalea leaf is used as an illustration in this case. The Bragg image produced clearly shows the veins of the vascular azalea leaf serving as a proof of concept for cost-effective potential application of acoustic Bragg imaging of biological objects in the medical field. Moreover, acousto-optic Bragg imaging is potentially harmless to biological cells and is sensitive to density and elastic variations in the tissue.

11:00

5aPA12. Acoustical activity of lithium niobate crystals. Farkhad R. Akhmedzhanov (Samarkand State Univ., 15 University Blvd, Samarkand 140104, Uzbekistan, farkhad2@yahoo.com)

Acoustical activity in Lithium Niobate crystals was investigated by method of Bragg light diffraction. Transverse acoustic waves were excited in the range of 0.4–1.6 GHz using quartz piezoelectric plates. $LiNbO_3$ samples were oriented at small angles to the axis *Z* in the crystallographic plane (010). Specific rotation of the polarization plane of transverse waves was determined from the dependence of the intensity of the diffracted light of the distance from the piezoelectric transducer along the acoustic wave propagation. The observed large specific rotation of the polarization plane of the acoustic waves can be used in devices controlling the intensity of light by the phenomenon of acoustical activity. Components of the imaginary part of the generalized Christoffel tensor taking into account the spatial dispersion have been determined also. It is shown that in crystals of class 3m the imaginary components of the Christoffel tensor for all the directions in the symmetry plane (100) are zero.

11:15

5aPA13. Reflected wave by the first interface of rigid porous medium at Darcy's regime. Mustapha Sadouki (Departement de Sci. de la matière, Université Djilali Bounaama à khemis-miliana, Rte. Thenia el Had, Ain Defla, Khemis-miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz), Zine El Abidine Fellah (LMA UPR7051 CNRS Aix-Marseille Univ, Centrale Marseille, Marseille, France), Mohamed Fellah (Fac. de Physique, USTHB, Alger, Algeria), and Claude Depollier (Laboratoire d'Acoustique de l'Université du Maine UFR STS, LUNAM Université du Maine, UMR CNRS 6613, Le Mans, France)

In this paper, reflection waves by the first interface of porous medium having rigid frame is considered. The method presented in this work is based on a temporal model of the direct problem in which a simplified expression of the reflection coefficient at the Darcy's regime (low frequency range) is established using equivalent fluid model, this expression depends only on the porosity and the viscous permeability (or the flow resistivity) of the medium. The simulated reflected wave is obtained in time domain by convolution between the reflected operator and the incident field. Experimental results are given for samples of air-saturated plastic foams and compared with theoretical predictions.

11:30

5aPA14. Characterization of rigid porous medium via ultrasonic reflected waves at oblique incidence. Mustapha Sadouki (Departement de Sci. de la matière, Université Djilali Bounaama à khemis-miliana, Rte. Thenia el Had, Ain Defla, Khemis-miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz), Amine Berbiche, Mohamed fellah (Fac. de Physique, USTHB, Alger, Algeria), Zine El Abidine fellah (LMA UPR7051 CNRS Aix-Marseille Univ, Centrale Marseille, Marseille, France), and Claude Depollier (Laboratoire d'Acoustique de l'Université du Maine UFR STS, LUNAM Université du Maine, UMR CNRS 6613, Le Mans, France)

In this paper, an enhanced method is proposed for measuring porosity, tortuosity, viscous, and thermal characteristic length of porous materials having a rigid frame via reflected ultrasonic waves at oblique incidence using the equivalent fluid model. The advantage of the proposed method is that the four parameters are determined simultaneously just using reflected experimental waves for a porous material saturated by air. The inverse problem is solved based on the least-square numerical method using experimental reflected waves in time domain. Tests are performed using industrial plastic foams. Experimental and numerical validation results of this method are presented.

Session 5aSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

Stephen Oregan, Chair
Naval Surface Warfare Center, Carderock Division

Contributed Papers

8:30

5aSA1. Active and passive techniques for a cost-effective noise reduction in the interior of aircraft cabins. Hasson M. Tavossi (Phys., Astronomy, and GeoSci., Valdosta State Univ., 2402, Spring Valley Cir, Valdosta, GA 31602, htavossi@valdosta.edu)

Different noise reduction techniques that can be applied to the interior of aircraft cabin are investigated, to determine the most cost-effective means, without any aircraft external modifications, such as jet engine design change, or fuselage modifications, with no significant added weight. The goal of this research is to arrive at a cabin interior design that can be retrofitted to the existing aircraft interior to reduce overall cabin noise. Relaxation oscillations of the aircraft cabin model, considered as a system in forced vibrations with non-linear damping, and sub-harmonic resonances are considered. Negative and positive damping coefficients, and some techniques for noise cancelation by active means are discussed. From noise power-spectrum results for a typical aircraft cabin, amplitude versus audible frequency, one can determine the most energetic cabin vibration modes. Those modes require the highest damping. The proposed technique will utilize a regular matrix of the multiple sets of open Helmholtz resonators, with sound absorbing surfaces, that are imbedded in the cabin interior walls, just below the surface, and tuned to the highest noise level frequencies inside the cabin. The resonators dissipate the noise energy inside the aircraft at the most dominant frequencies, and hence reduce the overall cabin noise level.

8:45

5aSA2. Numerical hybrid TMM-modal finite element method prediction of the vibroacoustic of sandwich panels with add-on damping. Imen Rzig, Noureddine Atalla, and Dilal Razhi (Mech., univesité de Sherbrooke, E3-2115, 2500 Boulvard de l'université, Sherbrooe, QC J1K2R1, Canada, imen.rzig@usherbrooke.ca)

This paper discusses the numerical modeling of the vibroacoustic response of sandwich-composite panels with add-on damping, under a diffuse acoustic field (DAF) excitation. A modal synthesis approach is used for the calculation of the structural response and the Rayleigh's integral is used for the acoustic response. Since the panel has a viscoelastic core, a methodology is presented to handle efficiently the modeling of the frequency depended properties of the viscoelastic layer. A hybrid TMM-modal FEM method is used to predict the acoustic response at high frequency, using the equivalent damping of panel which are calculated from strain energies. Next, a parameters study on the effect of the viscoelastic layer location is presented. In particular, three locations are compared: within the Honeycomb core, within the skins and added to the skin with a constraining layer. The effects of the excitation type on the vibration and acoustic response are also discussed. Key words: Sandwich NIDA, Modal FEM method, TMM method, viscoelastic damping, acoustic response, equivalent properties.

9:00

5aSA3. A modal solution for finite length rods with non-uniform area. Andrew J. Hull (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk derives a modal solution to the displacement field of a finite length rod whose area is varying with respect to its length. This method facilitates a solution to any problem where the area and derivative of the area can be represented as analytical functions. The problem begins by writing the longitudinal displacement of the non-uniform area rod as a series of indexed coefficients multiplied by the eigenfunctions of the uniform area rod. This series solution is inserted into the non-uniform area rod equation, multiplied by a single p-indexed eigenfunction and integrated over the interval of the rod. The resultant expressions can be written as a set of linear algebraic equations and this yields a solution to the displacement of the system. Five example problems are included: the first one has a non-uniform area that corresponds with a known analytical solution, the second has an area that can be represented by a Fourier series, and the third and fourth have areas that do not have a known analytical solution and the fifth is a generic second order non-constant coefficient differential equation. Four of these problems are verified with other methods. Convergence of the series solution is discussed.

9:15

5aSA4. Remote acoustic sensing of mechanical changes in thin vibrating plates. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Rm. 2010, Ann Arbor, MI 48109, tjayflyn@umich.edu)

Remote measurements of radiated sound from a vibrating structure may be exploited for non-contacting structural health monitoring. In this work, an experimental method for remote acoustic sensing of mechanical changes in thin vibrating plates is presented. The basic concept is to compare the radiated sound from baseline and mechanically modified plates to determine the presence or absence of mechanical modifications. Experimental results are presented for measurements of radiated sound from a 0.3-m-square by 1.6-mm-thick aluminum plate with clamped edges subject to swept-frequency shaker base-excitation in the bandwidth from 100 Hz to 4 kHz. The primary mechanical change considered in this investigation is added mass via magnets. Acoustic signals are collected with a linear array of sixteen microphones and are used to characterize the acoustic radiation from baseline and modified vibrating plates, to determine the thresholds above which mechanical changes can be reliably detected with statistical significance. This binary method of non-destructive evaluation can be extended to other classes of defects, such as holes, cuts (simulated cracks), and changes in the boundary conditions. With the integration of more advanced signal processing techniques, localization of the mechanical changes may be possible as well. [Sponsored by NAVSEA through the NEEC.]

9:30

5aSA5. The effect of elastic point contact: theory and experiment. Douglas Photiadis and David Goldstein (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, douglas.photiadis@nrl.navy.mil)

It has recently been pointed out that nearfield deformation in the neighborhood of an elastic point contact can have significant phenomenological effects, producing shifts of resonance frequencies and serving as a source of loss. In order to take these effects into account, it is necessary to employ three dimensional elasticity theory, precluding the use of lower dimensional dynamic models, for example, plate elements in a finite element analysis (FEA) computation. This is a very significant limitation because the frequency response of many large complex structures can only be predicted using such models. We propose here a theoretical modeling approach that resolves this problem and enables the use of lower dimensional models in the analysis of such systems. We have found the predictions of the model to be in good agreement with experimental measurements in the cases we have examined so far, translational and cantilever resonators. Research funded by the Office of Naval Research.

9:45

5aSA6. Calculation of energy exchange between the outer hair cell and structural components of the organ of Corti, using the *in vivo* measurement data. Amir Nankali and Karl Grosh (ME Dept., Univ. of Michigan, Ann Arbor, MI 48105, nankali@umich.edu)

The mammalian cochlea amplifies the sound born vibration of the microstructures of the OoC through a frequency and level dependent active process. The somatic motility of the mechanosensory outer hair cells (OHCs) is hypothesized as the key element of the cochlea active mechanism. During the sound stimulation, the OHCs respond to the membrane potential fluctuation through a fast alteration of their length. This cellular length change applies an active harmonic force to both sides of the OHC (reticular lamina (RL) on the apical part and basilar membrane (BM) on the basal part), and boosts the OoC motion. In this paper, we use the *in vivo* experimental data on the OHC extracellular receptor potential together with the displacement of the OoC structural components to estimate energy exchange between the modes. It is found that when the OHC transmembrane potential leads the BM displacement by a phase between 0 and 180 degree, with an optimal value of 90 degree, the electrical power is transmitted into the BM mode. A phase difference between 180 and 360 gives rise to power dissipation on the BM. These phase ranges are shifted by 180 degree for the RL mode. Analyzing the experimental data reveals that the OHC active force dissipate power in the BM side while it amplifies the RL motion. Moreover, we utilize the BM/RL displacement amplitude to quantify the OHC active power generated by each hair cell.

10:00–10:15 Break

10:15

5aSA7. Investigation of low-frequency broadband dipole transducer driving by PMN-PT transverse shear mode. Yuyu Dai, Yinqiu Zhou, Xiuming Wang, Hongbin He, and Zhengbo Wang (Inst. of Acoust., Chinese Acad. of Sci., No. 21, Bei-Si-huan-Xi Rd., Beijing 100190, China, daiyuyu001@126.com)

A low-frequency broadband dipole transducer with a standard dipole directivity pattern is a crucial factor in the research and development of a sonic logging tool. A new dipole transducer design based on cylindrical shell oscillating is proposed in this paper, in which transverse shear polarization PMN-PT crystal stacks are used to provide a driving force to excite a cylindrical-shell oscillation to generate dipole acoustic field. The resonant frequency of the transducer is decided by the dimensions of crystal stacks. Piezoelectric coefficient d_{15} of the PMN-PT crystals can reached 7000 pC/N, which means there is enormous potential in high power transducer design; Several ceramic plates are connected in a series to reduce the resonant frequency of the exciting source and increase the driving amplitude at the same time so as to enhance the radiation power further; several stacks of different sizes are connected in a parallel to broaden the bandwidth. Finite element method is adopted to optimize the structure of transducer. According to the optimized results, a prototype is produced and then tested in anechoic tank. Test results show that the transmitting response is larger than traditional dipole transducer and the directivity pattern is better in the operating frequency range.

10:30

5aSA8. Fluid-structure interaction effects for minimizing transmission in waveguides: Time and frequency domain approach. Swaroop R. Vettamparambath and Abhijit Sarkar (Mech. Eng., IndianInst. of Technol. Madras, #445, Pampa Hostel, Chennai, Tamil Nadu 600036, India, swarooprajvr@gmail.com)

In many noise and vibration control applications, transmission loss in waveguides needs to be maximized. We propose a waveguide comprising of two different fluids with large impedance mismatch. The two dissimilar fluids are separated by two identical spring-mass combinations. An analytical model for such waveguide is undertaken using principles of one-dimensional linear wave propagation theory. Transmission loss for the waveguide across the frequency range is formulated in terms of (i) impedance mismatch of the fluids (ii) fluid-structure interaction parameter. It is shown that an appropriate choice of the above two parameters leads to a minimal transmission across the frequency range. The above inference is also corroborated through transient Finite Element Analysis. For transient simulation an initial condition is imposed on the system and the simulation is carried on till the first transmission is observed. The ratio of the maximum response in the transmitted pulse to the maximum response in the incident pulse is defined as the transmission ratio in time domain. It is observed that the transmission ratio computed in time domain correlate well with the transmission ratio formulated in the frequency domain.

10:45

5aSA9. Railway vibration and noise reduction using particle impact dampers. Wonseok Yang (Mech. Eng., Hanyang Univ., Hanyang Univ., Haengdang 1-dong, Seongdong-gu, Seoul, Korea, Seoul ASI|KR|KS013| Seoul, South Korea, wonseok_yang@naver.com), SangKeun Ahn (Mech. Convergence Eng., Hanyang Univ., Seoul, South Korea), Hyoin Koh (Korea RailRd. Res. Inst., Seoul, South Korea), and Junhong Park (Mech. Convergence Eng., Hanyang Univ., Seoul, South Korea)

Rolling noise is becoming a increasing concern for residential areas near railways. Rail vibration caused by moving loads of a train and corrugations generates noise. This study presented a particle impact dampers and its application to reduction of the rolling noise. The particle impact dampers to efficiently reduce the vibration were designed and verified using simplified models of a railway. To verify the particle impact damper performance, the vibration of a simplified railway attached with particle impact dampers was measured. The effects of different clearances and mass ratio of the damper to the railway was investigated. The numerical predictions of finite element model of impacting dampers were proposed and verified to the measured results to find the vibration reduction mechanism.

11:00

5aSA10. On frequency dependences of material damping and dynamic elastic properties. Tamás Pritz (Budapest Univ. of Technol. and Economics, Műegyetem rkp. 3-9, Budapest 1111, Hungary, tampri@eik.bme.hu)

The various solid materials may exhibit diverse damping and dynamic elastic behavior as a function of frequency over a wide range extending from zero hertz up to the meaningful high frequencies. Nevertheless, some general characters in the dynamic behavior of materials can be experienced, especially if a limited frequency range is concerned. The aim of this paper is to establish and classify the general characters of frequency dependence of linear damping and elastic properties with special emphasize to the sonic range and the materials used for sound and vibration control. Based on simple physical views, the frequency dependences of damping properties (loss modulus and loss factor) are established as a first step. It is shown that basically two types of damping behavior can be distinguished, namely, (a) damping increasing with frequency and (b) damping exhibiting a peak. All other types of damping (e.g., the decreasing and the hysteretic damping) can be interpreted through the above mentioned ones. The dynamic elastic properties (shear or bulk modulus, etc.) as a function of frequency are determined from the damping by considering causality principle. The frequency dependences predicted for the dynamic moduli are discussed.

11:15

5aSA11. Determining optimal equivalent source positions in wave superposition method by self-adaptive searching algorithm. Shaowei Wu and Yang Xiang (School of Energy and Power Eng., Wuhan Univ. of Technol., Peace Ave., Wuhan, Hubei Province, No. 1040, Wuhan, Hubei 430063, China, thinkwsw@qq.com)

The calculation accuracy of wave superposition method is influenced greatly by equivalent sources, their positions, and the frequency. What's more, nonuniqueness occurs at eigenfrequencies when monopole or dipole is used as the equivalent source. Although tripole can overcome the nonuniqueness, the computational overhead is very large due to its complex expression and the accuracy is still affected greatly by the positions. A method is proposed to reduce the calculation errors. In this method, monopole, which is the simplest source, is used as the equivalent source. The upper limit frequency for a radiator under a certain meshing pattern is predicted by using the fictitious pressure generated via a reference source. Then, the optimal equivalent source positions for each frequency corresponding to the actual vibration velocity boundary condition, in which the average rate of pressure change is minimal, are determined within the upper limit frequency by a searching algorithm. Numerical simulation results of a complex structure show that the calculation errors are significantly reduced. At last, the method is verified by an experiment for a cuboid radiator. Experimental results show that the proposed method is practicable and good at accuracy.

11:30

5aSA12. Sound characteristic analysis of Multi-layer sound- absorbing materials using impedance tube. Hyungwoo Park (SoongSil Univ., 1212 Hyungham Eng. Building 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 156743, South Korea, pphw@ssu.ac.kr), Seonggeon Bae (Daelim Univ., Seoul, South Korea), Myungjin Bae (SoongSil Univ., Seoul, South Korea), and Duckhee Lee (Korea RailRd. Res. Inst., Gyeonggi-do, South Korea)

With the development of the field of transportation, vehicles have come to be driven fast. As a result, due to their high speed, railways are the cause of increased noise inside the railway vehicle. Methods for reducing noise present themselves in various ways. However, it is difficult for a method to be effective without a costly and difficult implementation of technology. In general, to improve the characteristics of vehicle interior noise conditions, which can increases the effect of sound absorbing or insulation materials, it is necessary that any sound absorbing material be located between the outer and inner walls of the vehicle. In this paper, we perform an experiment on the acoustic characteristics of a composite multilayer type sound-absorbing material that can be used to reduce the interior noise of a vehicle. In our tests, we used an impedance tube, which can scan a plane wave with a speaker for the test-material. The tube is fabricated in a circular symmetrical shape. This overall method can be used to evaluate acoustic transmission loss. The results found in this study can be used to show that the selection of a multi-layer sound-absorbing material can allow users to achieve the characteristics of noise reduction. With the results delineated in this study, a multi-layer high density material can be confirmed to show improved performance this material can also be used to check the characteristics of the frequency response of the shape of the surfaces.

FRIDAY MORNING, 6 NOVEMBER 2015 GRAND BALLROOM 8, 8:30 A.M. TO 10:00 A.M.

Session 5aSCa

Speech Communication: Intonation, Tone, and Prosody (Poster Session)

Tessa Bent, Chair
Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405

Authors will be at their posters from 8:30 a.m. to 10:00 a.m. To allow authors an opportunity to see other posters in their session, all posters will be on display from 8:30 a.m. to 12:00 noon.

Contributed Papers

5aSCa1. Two strategies for distinguishing ngã and sắc tones in Northern Vietnamese. Taylor L. Miller, Angeliki Athanasopoulou, Nadya Pincus, and Irene Vogel (Linguist & Cognit. Sci., Univ. of Delaware, 125 E Main St., Newark, DE 19716, tlmiller@udel.edu)

The six tones of Northern Vietnamese involve F0 and phonation properties. We examine the acoustic manifestation of two rising tones usually characterized as having distinct phonation (ngã = creaky and sắc = modal) in 1584 vowels produced by 9 Hanoi speakers (88 real three word compounds, 8 target vowels /a/, /i/, /u/ with sắc and ngã in first two syllables). Based on measurements of F0, energy, duration, and phonation properties (spectral tilt, CPP, and HNR), we observed two strategies for producing the two tones: (a) both F0 and phonation differences, where creaky voice appeared in >78% of the ngã tones (N = 7); (b) only F0 difference, where creaky voice appeared in <6% of the ngã tones (N = 2). Classification of the data into the two tones with Binary Logistic Regression Analyses confirmed the distinct behaviors. In the first strategy, the main property distinguishing ngã from sắc is HNR (84%), but F0 was also very successful (75%). In the second strategy, F0 was the only significant property (90%). Given that there were no age, gender, or educational differences, we suggest that the patterns may be due to (i) a regional dialectal difference or (ii) a change in progress in Vietnamese.

5aSCa2. Perception and production abilities of question vs. statement intonation patterns in young deaf children with early cochlear implantation. Sangsook Choi, Ioana Barbu, Cynthia Core, and James Mahshie (Speech and Hearing Sci., The George Washington Univ., 2115 G St. NW, Washington, DC 20052, sangsook_choi@email.gwu.edu)

Little research exists studying production and perception of prosody in early-implanted children with cochlear implants (CIs). Because of limited F0 information conveyed by CIs, there is particular interest in the impact of CIs on intonation perception and production. The aim of the present study was to describe and compare the ability of hearing (HC) and implanted (IC) children to perceive and produce distinctions between question vs. statement intonation patterns. Nine HC and 9 IC between 38 and 58 months of age participated in the study. All IC received their first implant prior to 15 months and had no other identified medical or developmental problems

5a FRI. AM

besides hearing loss. To determine the children's ability to perceive rising and falling intonation, an imitation task was used. Results showed no significant difference in perception between the groups ($p = .09$). Intonation production was examined by eliciting yes–no questions and statements during a play-based task. Production measures included mean, range, and slope of F0. While F0 range was significantly greater for IC than for HC ($p = .015$), no other measure was found to be significantly different between the two groups. Overall, early-implanted deaf children with CIs appear to develop intonation comparable to their hearing peers.

5aSCa3. Speech variability and prosody in childhood apraxia of speech. Toby Macrae, Kaitlin Lansford, and Emily Berteau (Commun. Sci. and Disord., Florida State Univ., 201 W. Bloxham St., Tallahassee, FL 32306-1200, tmacrae@fsu.edu)

This is a follow-up to a paper that was presented at the 2014 Convention of the American Speech-Language-Hearing Association, which included preliminary data from 12 children. Data collection is ongoing and the data from all participants will be presented at the Acoustical Society of America Meeting in 2015. While speech variability and disordered prosody are core features of childhood apraxia of speech (CAS), much of the research in this area is descriptive and/or subjective. The purpose of the proposed study is to compare children with CAS to children with non-CAS speech sound disorders (SSDs) using more objective acoustic measures of variability and prosody, including: (1) durational variability of the phrase "Buy Bobby a puppy," (2) durational variability of /ɑ/ in "Bobby," (3) durational variability of /ʌ/ in "puppy," (4) durational variability of voice onset time for /p/ in "puppy," (5) spectral variability of the first and second formants (F1 and F2) for /ɑ/ in "Bobby," (6) spectral variability of F1 and F2 for /ʌ/ in "puppy," and (7) three stress metrics from Liss *et al.* (2009), which relate variability in consonant durations to variability in vowel durations and have been shown to differentiate between speakers with and without dysarthria and among different dysarthria subtypes. Preliminary data were encouraging. Children with CAS showed relatively more variability in consonant durations than vowel durations. Children with non-CAS SSDs showed relatively more variability in vowel durations than consonant durations.

5aSCa4. The intonation of wh-in situ questions in Northern Peninsular Spanish. Carolina Gonzalez and Lara Reglero (Modern Lang. and Linguist, Florida State Univ., 625 University Way, DIF 322, Tallahassee, FL 32306-1540, cgonzalez3@fsu.edu)

This project investigates the intonational characteristics of wh-in situ questions (As in: ¿Compraste qué? *'You bought what?'*) in Northern Peninsular Spanish. Although these questions reportedly occur with various pragmatic readings in several Spanish dialects (Uribe-Etxebarria 2002; Chernova 2013; Reglero and Ticio 2013), their intonation remains to be investigated. Data from 22 speakers from Northern Peninsular Spanish was collected in a reading and an elicitation task. Each task included 30 contextualized wh-in situ questions representing information-seeking, echo-repetition, and echo-surprise readings. Target sentences were interspersed among declaratives and yes–no questions. Preliminary results from the elicitation task are discussed for two female participants. Prevalent pitch contours are analyzed following Spanish ToBi conventions (Face and Prieto 2007). In addition, we report the global tonal range for each type of wh-in situ, as well as the tonal range average from the nuclear tone to the boundary tone. It is expected that surprise contexts will have the greatest tonal range, followed by echo-repetition contexts. Finally, we explore the prevalence of intermediate phrases prior to the final wh-word for the three contexts, which might have repercussions for the syntactic analysis of wh-in situ in Spanish (Uribe-Etxebarria 2002 vs. Reglero and Ticio 2013).

5aSCa5. The intonation of declaratives and absolute interrogatives in Valencian Spanish. Jessica Craft (Spanish, Florida State Univ., 625 University Way, DIF356, Tallahassee, FL 32306, jmc07f@my.fsu.edu)

This study presents data on the intonation of broad focus declarative sentences and information-seeking absolute interrogative sentences in Valencian Spanish, as the characteristics of this dialect have not previously been described. The main research questions of this study are what types of intonation patterns are typical for Spanish in this region and in what ways these patterns are similar and/or different from the ones attested for Castilian Spanish and Valencian Catalan. The speech data analyzed came from six native speakers of Valencian Spanish, all early bilinguals of Valencian Catalan. A total of 240 sentences were analyzed in Praat (Boersma and Weenink 2014) following standard Sp_{ToBI} and Cat_{ToBI} conventions (Beckman, *et al.* 2002; Estebas-Vilaplana and Prieto 2010; Prieto 2014). The study focuses on the pitch accents used in the prenuclear and nuclear positions, as well as the boundary tones for both sentence types. The speech data show use of intonational contours from both Valencian Catalan and Castilian Spanish and demonstrates the profound influence of Catalan in the Spanish of this region. Furthermore, the data are consistent with what has been reported for other Spanish-Catalan bilingual speakers in other regions of Spain (Romera and Elordieta 2013, Simonet 2011, Romera, *et al.* 2008).

5aSCa6. Production of emotional intonation among Mandarin and English speakers. Ratree Wayland, Yiqing Zhu, and Michelle Perdomo (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu)

Some researchers claim that intonation can be used to express specific emotion while others argue against the existence of emotion specific intonation patterns. In addition, languages differ in their use of intonation pattern to deliver similar emotion, and that L2 learners have the tendency to use L1 knowledge to produce intonation. Previous research shows that a falling successive addition boundary tone was used to express "disgust" or "anger" while a rising successive addition tone was used to convey "surprise" and "happy" emotions in Mandarin. In this study, we compare intonation patterns used to express five emotions: anger, disgust, surprise, joy, and neutral by 10 Mandarin and 10 English speakers in 1, 2, or 5-word utterances in English. Mandarin speakers were also asked to produce all 5 emotions in 1, 2 and 5-word utterances in Mandarin. Preliminary analyses from one Mandarin speaker showed that mean F0 of utterances produced with different emotions are significantly different in all three utterance lengths in both Mandarin and English. Inconsistent with previous research, a "falling" successive addition tone is used in all five emotions in Mandarin and in four emotions, except disgust, in English.

5aSCa7. L + H* and H* pitch accents in Mandarin Chinese. Yiqing Zhu and Ratree Wayland (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu)

In this study, we examine acoustic correlates of L + H* and H* pitch accents in Mandarin Chinese. In English, L + H* denotes "exclusiveness" and H* conveys "new" information. For example, L + H* A. Katie did not pet the cat, (Kellie did). H* B. Katie did not pet the cat (she pet the dog). Ten Mandarin speakers were asked to produce the English equivalent L + H* and H* pitch accents in Mandarin as in C and D: L + H* C. 咪妮 没有 摸 猫, 妞蒙 摸 了 猫 MiNi meiyou mo mao, NiuMeng mo le mao. *MiNi did not pet the cat, it is NiuMeng who petted the cat.* H* D. 咪妮 没有 摸 猫, 咪妮也 没有 摸 狗 MiNi meiyou mo mao, MiNi ye meiyou mo gou. *MiNi did not pet the cat, and she did not pet the dog either.* MiNi and NiuMeng were produced with all combinations of four Mandarin tones. Preliminary findings from one speaker suggest an interaction between tones and pitch accents such that pitch contour and pitch height of both types of pitch accents vary as a function of tones.

5aSCa8. Durational characteristics of sentence-medial and sentence-final pauses in the production of a paragraph. Kuniko Kakita (Liberal Arts and Sci., Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, kakita@m3.spacelan.ne.jp) and Shizuo Hiki (Faculty of Human Sci., Waseda Univ., Tokorozawa, Japan)

The present study investigated the durational characteristics of sentence-medial and sentence-final pauses in the production of a paragraph, with an aim to elucidate the factors that determine the durational organization of connected speech. A phonetically trained native speaker of Japanese read a paragraph consisting of eight sentences at a moderate speaking rate. Three recordings varying slightly in total duration — roughly 55 s, 52 s (−5%), and 49s (−10%) — were analyzed acoustically. Results showed that approximately 30% of the paragraph duration was pause duration, of which about 80% was sentence-final and 20% sentence-medial. Pause duration affected paragraph duration more significantly than speech duration, e.g., 10% shortening in paragraph duration resulted from 25% shortening in pause duration but only 5% shortening in speech duration. The duration of sentence-medial pauses was positively correlated to the duration of preceding speech, while the duration of sentence-final pauses reflected the internal structure of the paragraph, i.e., paragraph-internal topic transitions were accompanied by longer pauses, possibly providing opportunities, too, for physiological adjustments such as breathing and swallowing. The ratio of accumulated pause duration to accumulated speech duration increased in an asymptotic manner, approaching the final ratio value near the end of paragraph production.

FRIDAY MORNING, 6 NOVEMBER 2015 GRAND BALLROOM 8, 10:30 A.M. TO 12:00 NOON

Session 5aSCb

Speech Communication: Foreign Accent and Multilingual Speech Production and Perception (Poster Session)

Rajka Smiljanic, Chair
Linguistics, University of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

Authors will be at their posters from 10:30 a.m. to 12:00 noon. To allow authors an opportunity to see other posters in their session, all posters will be on display from 8:30 a.m. to 12:00 noon.

Contributed Papers

5aSCb1. Acoustic detail in monolingual and bilingual children's representations of English and Spanish. Cynthia P. Blanco, Rajka Smiljanic (Univ. of Texas at Austin, 305 E. 23rd St., Linguist, B5100, Austin, TX 78712, cindyblanco@utexas.edu), and Colin Bannard (Univ. of Liverpool, Liverpool, United Kingdom)

Children must learn to process variations in the pronunciation of their language(s), but they must also learn the social meaning of particular kinds of variation. By preschool, children's initially overly specific lexical representations have generalized to reflect commonalities in pronunciations, thus allowing children to recognize accented productions as instances of familiar lexical items (Best *et al.*, 2009; Schmale, *et al.*, 2010, 2011, 2012; Stager and Werker, 1997). However, elementary-school-aged children struggle to use the phonetic variation present in accented speech to understand social differences among talkers (Floccia *et al.*, 2009; Girard *et al.*, 2008). In the present study, we investigated children's ability to associate acoustic cues with a particular language as a function of their language background (monolingual, significant L2 exposure, or bilingual). Children decided whether nonce words containing language-specific sounds were produced by a Spanish speaker or an English speaker. Language-specific cues included phonemes unique to Spanish (/r/) or English (/ɹ, θ/), or sound categories common to both languages but produced differently in each (/l, u/). The categorization patterns and reaction times of the three groups were compared, and results indicate that children with exposure to or proficiency in Spanish categorized the nonce words more accurately than monolinguals.

5aSCb2. Phonetic divergence in bilingual speakers is modulated by language attitude. Wai Ling Law (Linguist, Purdue Univ., Beering Hall of Liberal Arts and Education, 100 North University St., West Lafayette, IN 47907, wlaw@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Bilingual speakers' speech varies phonetically according to many factors such as age of arrival and of acquisition of the second language (L2) (e.g., Flege *et al.*, 1999). However, such temporal factors may not be as relevant to populations with more uniform language experience, as in pervasively multilingual societies. In such diglossic situations, speakers' attitudes toward each of their languages may have a stronger influence on everyday pronunciation, as they do for language learning (Moyer, 2007) and phonetic accommodation to interlocutors (Dmitrieva *et al.*, 2015). This study was designed to determine whether bilingual speakers' attitude toward their language(s) and associated culture(s) modulates phonetic properties of their speech. Native Cantonese-English bilinguals living in Hong Kong ($N = 20$) produced near homophones in both languages under conditions emphasizing each language on different days. The degree of diphthongization of Cantonese /o/ and English /oʊ/, and Cantonese /aɪ/ and English /ɑɪ/ were quantified acoustically and compared to attitude scores elicited in a questionnaire. Participants with a more positive attitude toward Cantonese showed larger cross-language differences in diphthongization of these pairs, demonstrating an effect of language attitude on phonetic properties of speech and highlighting the contribution of sociolinguistic factors to phonetic variability in diglossic contexts.

5aSCb3. How do L1 and L2 influence on the acquisition of L3 (English) stress pattern. Mahire Yakup and Dina Omanova (World Lang. and Lit., Nazarbayev Univ., Block 38-1104, 53 Kabanbay Batyr Ave., Astana, Aqmola 010000, Kazakhstan, yakefu.mayila@nu.edu.kz)

In this research, we investigated the acquisition of English stress by trilingual speakers (Kazakh/Russian (L1)-Russian/Kazakh (L2)-English (L3). Stress in Kazakh, as a Turkic language, is on the final position (Johnson, 1998) and was cued by fundamental frequency (Kirghner, cited from Kondibaeva, 2010). However, in Russian, stress was cued by duration and intensity (Hamilton, 1980; Kuznetsova, 2006). In this research, we used two different trilingual groups in which the Kazakh-Russian-English trilingual group has Kazakh as a dominant language and first language; on the other hand, the Russian-Kazakh-English trilingual group has Russian as a dominant language with the advanced level of Kazakh. However, both groups have high level of English (IELTS = 6.5 and above). All participants from both groups produced the noun-verb stress pattern words in sentences. In the production of ten female speakers from each group, average fundamental frequency, duration, average intensity, and first and second formant frequencies for vowels were collected in the stressed and unstressed syllables. The result showed that for Russian-Kazakh-English speakers, duration and

intensity are stronger cues than F0. Kazakh-Russian-English trilinguals used all parameters, but duration was strongly associated with final lengthening. The results will be discussed in terms of L1/L2 transfer into the acquisition of L3.

5aSCb4. Accent and fluency in third language acquisition. Hiromi Onishi (Grinnell College, 1210 Park St., Grinnell, IA 50112, honishi84@gmail.com)

This paper examined the influence of L2 on L3 accent and fluency. Target participants were native speakers of Chinese who have studied English as an L2 prior to beginning their study of Japanese as L3. Participants were recorded in L2 and L3, and their speech samples were rated by expert judges of respective language. The judges were asked to rate each speaker based on foreign accentedness and overall fluency. The results from English and Japanese were analyzed using correlation analysis. The result of the correlation analysis showed a positive correlation between accent in L2 and L3, which suggests that, regarding foreign accentedness, the better a speaker performs in L2, the better they also perform in L3. The correlation between L2 and L3 in terms of fluency did not reach significance, mostly due to the fact that fluency rating in English was done mildly by the judges. The result nevertheless showed a clear trend toward positive correlation between the two languages. This study contributes to the area of Third Language Acquisition by supporting the idea that L3 acquisition is qualitatively different from L2 acquisition, and phonological acquisition in L3 is influenced not only by the learner's L1 but also their L2.

5aSCb5. Foreign accentedness of English sentences spoken by Japanese EFL learners and Japanese teachers of English: A first report. Natsumi Maeda (DaitoBunka Univ., 1-9-1, Takashimadaira, Itabashi, Tokyo 175-8571, Japan, h20625maeda@hotmail.co.jp) and Kiyoko Yoneyama (Daito-Bunka Univ., Itabashi-ku, Tokyo, Japan)

This study reports the results of a foreign-accented-rating experiment that investigate the foreign accentedness of spoken English sentences by two Japanese groups, Japanese EFL learners and Japanese teachers of English. This study aims first to investigate whether spoken English sentences by Japanese teachers of English are judged less foreign-accented than those by Japanese EFL learners, and second to investigate whether American-English listeners rate spoken English sentences by Japanese speakers more severely than Japanese listeners do. The stimuli were five sentences adopted from Flege, Munro, and McKay (1995) spoken by 33 Japanese EFL learners and 33 Japanese teachers of English. Ten American-English speakers and ten Japanese speakers were asked to rate the stimuli presented visually and auditorily by clicking along a line on the computer screen for their ratings. The participants' original responses were converted to 10 scales and were submitted to the analyses. The results showed that spoken English sentences by Japanese teachers of English were rated significantly less Japanese-accented than those by Japanese EFL learners. The results further revealed that as in the previous studies, the American-English speakers rated spoken English sentences by two Japanese groups significantly more severely than Japanese speakers. [Work supported by JSPS.]

5aSCb6. Non-word repetition task by young Japanese learners of English. Hiromi Kawai (Ctr. for Teaching English to Children, Kanda Univ. of Int.. Studies, 1-4-1 Wakaba, Mihama-ku, Chiba City 2610014, Japan, kawai-h@kanda.kuis.ac.jp)

This study identified the L2 sound processing problems of young Japanese EFL learners on articulatory production ability of English sounds, without intervention of lexical representation. A non-word repetition test was conducted based on the L1 English children's speech processing system as proposed by Stackhouse and Wells (1997). 203 5th and 6th graders from a public elementary school in Tokyo participated in this study. Nine L1 English normally developing children participated in the test as a reference group. During the non-word repetition test, the participant children watched a native speaker of American Standard English on a computer monitor pronouncing 15 non-words ranging in length from one to three syllables and repeated each non-word following the NS model pronunciation. The recorded performances of the participants were judged by two experts in phonetics and phonology and a native speaker of English specializing in university-TESOL. There was no significant difference between the three judges in perceptual rating ($p > .05$). An ANOVA showed a significant difference between the Japanese children and the L1 English children in the total score of the repetition test. The difficulty of the item analysis, however, revealed a tendency in both Japanese and L1 English children in articulating each syllable.

5aSCb7. Listeners pay attention to rhythmic cues when judging the nativeness of speech. Elisa Pellegrino (Dept. of Literary, Linguist and Comparative Studies, Univ. of Naples L'Orientale, via Duomo 219, Naples 80138, Italy, pellegrino.elisa.1981@gmail.com) and Volker Dellwo (Univ. of Zuerich, Zurich, Switzerland)

Native listeners are good at detecting whether speech is foreign-accented or not. Here, we tested the role of rhythmic cues from the amplitude envelope (ENV) in this process. In a binary forced-choice perception experiment, ten L1 Italian listeners listened to 32 stimuli, each containing two Italian utterances of identical lexical content under the following conditions: (a) Both utterances were produced by a non-native speaker and were manipulated with either the ENV of a native German speaker of Italian and Italian L1 speaker (at 10 and 30 frequency bands). (b) One utterance was produced by an L1 Italian, the other by an L2 speaker. For 10 and 30 frequency bands their ENVs were exchanged (speech chimeras). Listeners' task was to choose which of the utterances in a stimulus was the more native-like. In condition (a), listeners' probability to choose the utterance with the L1 envelope was above chance at 10 bands (0.72) and increased with 30 bands (0.78). In condition (b) listeners' probability to choose the utterance with the L1 ENV was 0.19 at 10 bands and 0.46 at 30 bands. We conclude that rhythmic cues in the speech ENV influence listeners' perception of nativeness.

5aSCb8. Disentangling the contribution of pitch and duration cues in first and second language perception of the Mandarin neutral tone. Arthur L. Thompson and Francis Nolan (Dept. of Theor. and Appl. Linguist, Univ. of Cambridge, Sidgwick Ave., Cambridge CB3 9DA, United Kingdom, alt54@cam.ac.uk)

Most L1 and L2 Standard Mandarin (SM) perception tests focus on the four lexical tones. However, none take into account the neutral tone (NT) for L2 speakers, and few do for L1 speakers (e.g., Yang, 2010). Pitch onset aside, the relatively short duration of NT syllables is assumed to be their primary perceptual cue (Lin and Yan, 1980). This study tested which perceptual cues are used by L1 and L2 groups. To this end, 9 L1 and 9 L2 learners (L1 British English) participated in a five-way alternative forced-choice tone identification task. Three minimal pair disyllabic tokens containing NT were used as stimuli. Syllables carrying NT were manipulated in PRAAT using steps of 30, 50, 70, and 100% manipulation for pitch and/or duration. Pitch was progressively manipulated to mimic that of a lexical tone, while duration progressively approached that of a lexical tone in second syllable position. This study found that L2s rely primarily on duration to identify NT, while L1s rely primarily on pitch. This expands previous studies showing that duration interferes with L2 SM tone perception (Blicher *et al.*, 1991; Chang, 2011). By contextualizing this finding, new models are proposed for L2 SM tone perception and acquisition.

5aSCb9. Production and perception skill developments of Korean coronal obstruents by inexperienced English-speaking learners of Korean. Hanyong Park (Dept. of Linguist, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53211, park27@uwm.edu)

The present study investigates how speech production and perception abilities develop in second language learning among learners in a classroom setting. English-speaking students taking a first semester college course in Korean participated in both production and perception tasks. In the production task, the participants read a list of Korean frame sentences with target stimuli consisting of Korean /t t' t^h s s'/ with /a/ in CV and VCV. To assess the production accuracy, native Koreans identified the consonants from the learners' productions. In the perception task, the participants listened to the same corpus of Korean stimuli and identified the consonants with Korean orthography. The data were collected four times over two semesters: Week 5 or 6, and Week 13 or 14 of each semester. Results indicate that the learners' overall production and perception abilities are correlated to each other in some but not all data collection times. Further analyses show that over two semesters, most learners did not improve much in overall perception abilities. Not much learning occurred either in production or perception for some sounds (e.g., /s/, /s'/). These findings suggest different learning units for production and perception as well as the powerful influence of native language.

5aSCb10. Speaking rate variability in spontaneous productions by non-native speakers. Tuuli Morrill (Linguist, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, tmorrill@gmu.edu) and Melissa Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR)

Most research examining differences between native and non-native speech measures mean differences at the segmental level. Some work has examined non-segmental characteristics of speech such as speaking rate; however, this work has also typically examined only mean differences. In the present study, we ask whether within-speaker variability, in addition to mean differences, characterizes differences between non-native and native speech. Specifically, we examine speaking rate in spontaneous productions by both native and non-native speakers of English. Preliminary work suggests that rate change across utterances in read productions of non-native speech is more variable than in read productions of native speech (Baese-Berk and Morrill, 2014, *Indianapolis ASA*). However, it is possible that read speech contains sources of variability that are specific to processing difficulties during reading. In the present study, we examine native speakers of Korean and Mandarin producing spontaneous speech in English and compare their speech to both native speakers of English, and read productions by these same non-native speakers. We measure mean speaking rate within utterances, as well as the amount of rate change (slowing or speeding up of speaking rate) from utterance to utterance. Results will contribute to an understanding of the role of variability in non-native speech production.

5aSCb11. Effects of training methods and attention on the identification and discrimination of American English coda nasals by native Japanese listeners. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

The accuracy with which native Japanese listeners identified and discriminated American English coda nasals in /CVN/ context was assessed before and after training. The listeners were divided into four groups, each of which received a different type of training. Two of the four groups were vowel-oriented; one of these groups received vowel identification training (VI), while the other received vowel discrimination training (VD). The other two groups were nasal-oriented. One of the nasal-oriented groups received nasal identification training (NI), and the other received nasal discrimination training (ND). The results revealed that the two nasal-oriented groups made more gains in its ability to identify and discriminate American English coda nasals than the vowel-oriented groups after training. The result implies that identification and discrimination trainings are equally effective in improving listeners' sensitivity to identify and discriminate American English coda nasals. The two vowel-oriented groups achieved modest improvement in identification and discrimination accuracy, which suggests that repeated exposure to stimuli can enhance listeners' sensitivity even when their attention is not on the target segment.

5aSCb12. One of these accents sounds like the other; one of these accents is not the same. Rachel M. Miller (Psych., California State Univ. San Marcos, 333 S. Twin Oaks Valley Rd., San Marcos, CA 92078, rmiller@csusm.edu)

Accented speech occurs when the structure of a talker's native language causes deviations from the speech productions norms of the non-native language being produced (e.g., Spanish-accented English; Tarone, 1987). The current study tested sensitivity to deviations shared across accents by asking two groups of listeners to judge similarities and differences between Spanish- and Chinese-accented speech samples. The matching group was asked to judge whether a model's accented token (e.g., Spanish) was more similar in accent to a token in the same (e.g., Spanish) vs. a different accent (e.g., Chinese). The discrimination group was asked to judge whether a model's accented token (e.g., Spanish) was different in accent from a token in the same (e.g., Spanish) vs. a different accent (e.g., Chinese). Results showed that listeners are able to make similarity matches at significantly greater than chance (50%) levels, suggesting that they are perceptually sensitive to the similarities between non-native accents. However, listeners were not good at picking out the "dissimilar" accent. This experiment shows a discrepancy in listeners' ability to compare accents based on whether they focus their judgments on similarities or differences between the accents, suggesting that different cognitive processes may be used in matching versus discrimination type tasks.

5aSCb13. The effect of informational and energetic masking on foreign-accent adaptation. Elisa Ferracane, Cynthia P. Blanco, Michelle Dubois, Andrea Manrique, and Rajka Smiljanic (Linguist, Univ. of Texas, 305 E. 23rd St., Austin, TX 78712, elisa@ferracane.com)

Speech understanding in noisy environments can be compromised through energetic (EM) and informational (IM) masking. EM reduces intelligibility of target speech through spectro-temporal overlap with the masker at the auditory periphery level. IM refers to the higher-level interference, such as competing attention, linguistic interference, and increased cognitive load. The present study examined the effects of EM and IM on foreign-accent adaptation. Native English listeners heard blocks of sentences produced by native-accented (NA) or foreign-accented (FA) talkers (Korean, Spanish) mixed with speech-shaped noise (SSN) or two-talker, native-accented babble and responded to a visual probe. Preliminary results show that listeners were more accurate and faster in babble compared to SSN. The more successful FA adaptation in the babble condition may be related to the presence of dips in the masker energy or the ability of listeners to successfully separate FA target from the NA background speech. Additionally, the acoustic properties of the target sentences, such as speaking rate and pausing, are examined to understand their effect on FA adaptation in each of the two noise conditions. These findings suggest that IM was less disruptive than EM for FA adaptation even though listeners were processing FA speech with an increased cognitive load.

5aSCb14. The effect of background speech variation on perceived foreign accent. Dylan Pearson (Linguist, Univ. of Wisconsin-Milwaukee, 3243 N Downer Ave., Milwaukee, WI 53211, dvp@uwm.edu), Amara Sankhagowit (Linguist, Univ. of Chicago, Milwaukee, WI), and Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

We investigate what effect the presence of differing degrees of accented babble speech has on the perceived foreign accent of the target utterances. We presented various degrees of foreign accented English sentences in multi-talker babble conditions to native speakers of English to rate the accentedness on a 1−9 Likert scale. Male and female native speakers of English and native Korean speakers having light to heavy foreign accent produced the target stimuli. We overlapped babble recorded from two talkers (1 male and 1 female) to generate the multi-talker babbles, where the talkers were speaking in native English, native Korean, heavy accented English, and light accented English. Our results show a tendency for listeners to rate the stimuli presented in more native-rated babble as sounding more native-like than the same utterance presented with less native babble. The data also points to native target speech being more sensitive to a change in the degree of accent in babble than non-native target speech. All taken together, the results suggest a "bleedthrough effect" as a result of pieces of the background babble appearing to be processed along with the target speech, thus affecting the perceived accent.

Session 5aUW

Underwater Acoustics: Communications, Transducers, Target Response, and Nonlinear Acoustics

Raymond Lim, Cochair
Naval Surface Warfare Center Panama City Division, 110 Vernon Ave., Code X11, Panama City, FL 32407-7001

Simon E. Freeman, Cochair
Scripps Institution of Oceanography, 7038 Old Brentford Road, Alexandria, VA 22310

Contributed Papers

8:00

5aUW1. A single equation for predicting required ocean and seismic acoustic fiber optic sensor response. Fred C. DeMetz (Sound Path Technologies, LLC, 9056 Camellia Ct, Ste. 100, Rancho Cucamonga, CA 91737, fred@soundpathtech.com)

In the past decade, requirements have evolved for systems of ocean bottom and seismic downhole acoustic sensors with high dynamic range, wider bandwidths, and which will operate reliably for a decade or more in high-pressure-high-temperature environments. As seismic and acoustic survey spatial resolution, operating depth, and range requirements have increased, higher sensor string counts have been required to provide the needed improvements in target definition. Fiber optic sensors and telemetry systems are being developed which allow HPHT operation, without vulnerable electrical telemetry in the "wet" end of the system, and which can reliably transmit high-sensor count string outputs which are 10's of km from the processor. This paper introduces a modified version of the SONAR equation which allows straight forward estimation of the required fiber optic acoustic sensor response, early in the system design stage, and allows the system engineer to define and optimize sensor response requirements and help minimize system performance risks.

8:15

5aUW2. Wireless underwater acoustic beamforming using chip-scale atomic clock timers. Simon E. Freeman (Underwater Acoust., Naval Res. Lab., 6819 Duke Dr., Alexandria, VA 22307, simon.freeman@gmail.com), Lloyd Emokpae, and Geoffrey F. Edelmann (Underwater Acoust., Naval Res. Lab., Washington, DC)

Obtaining the directionality of the sound field is typically performed using a cabled array of hydrophones, connected to a central data collection device. The cable introduces vulnerability wherein an electrical break in the line or dragging by currents or anchors may render the array inoperable. An underwater wireless beamforming system, which transmits time and recorded low-frequency sounds using directional high-frequency acoustic communications between elements, is presented here. Each element is contained in a standalone housing that contains batteries, amplifiers, signal conditioning hardware and a field-programmable gate array (FPGA)-based signal processing computer. Low-frequency sounds are recorded at a sampling rate of 2 kHz, then transmitted using binary phase-shift-key (BPSK) encoding at a center frequency of 750 kHz. Inter-element spacing estimates are facilitated by the time-stamped high-frequency signals. System timing is controlled by a Symmetricom® SA.45s chip-scale atomic clock (CSAC) embedded in each unit, connected to the FPGA. In addition, a CSAC output waveform is divided and modified to form the high-frequency multiplier waveform for the BPSK signal. The elements were deployed in shallow water offshore of Panama City, FL, in August 2015. [This work was supported by the Office of Naval Research.]

8:30

5aUW3. Improved multiple focusing with adaptive time-reversal mirror in the ocean. Gi Hoon Byun and Jea Soo Kim (Ocean Eng., Korea Maritime and Ocean Univ., Korea Maritime Univ., Dongsam 2-dong, Yeongdo-gu, Busan 606-791, South Korea, gihoonbyun77@gmail.com)

The linearly constrained minimum variance (LCMV) method in adaptive signal processing is designed in such a way that it attempts to minimize the output power subject to constraints on the look directions. This adaptive method has been applied to time-reversal mirror (TRM) for simultaneous multiple focusing and its efficiency has been verified by [Kim*et al.*, J. Acoust. Soc. Am. **109** (5), 1817–1825 (2001)]. However, the norm of the weighting vector $\|\mathbf{w}\|$ that satisfies the constraints tends to become very large when two probe sources (PS) are close to each other. It causes prominent spatial sidelobes as the weighting vector $\mathbf{w}$ is back-propagated in TR focusing. In this study, a relationship between $\|\mathbf{w}\|$ and PS locations is illustrated, and LCMV method is partially reformulated to calculate the weighting vector $\mathbf{w}$ which satisfies new constraint responses, so that sidelobes are significantly suppressed. The proposed method for improved adaptive time-reversal mirror for stable simultaneous multiple focusing is verified using numerical simulations and experimental data.

8:45

5aUW4. Focal properties of an underwater acoustic Fresnel zone plate lens. David C. Calvo, Michael Nicholas, Abel L. Thangawng, and Christopher N. Layman (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

Fresnel zone plate (FZP) lenses provide thin alternatives to conventionally sized acoustic lenses. We present experimental characterization of an underwater acoustic FZP lens fabricated at the Naval Research Laboratory. This diffractive lens has a short focal length (numerical aperture angle of $= 62°$) which provides a narrow focal spot with a diffraction limited first-null radius 0.61λ/sin. The 13 in diameter FZP consists of opaque rubber foam rings attached to a thin PDMS rubber substrate. Ultrasonic measurements of transmission were made using a scanning needle hydrophone and a 200 kHz piston source. Measured focal gain varied between 20 and 15 dB for incidence angles between 0 and 12°, respectively. These gains agreed with predictions computed using a wide-angle beam propagation method. Measured transverse focal shift agreed with geometrical constructions used in conventional lens theory. Axisymmetric finite-element computations using Fourier decomposition of the oblique incident field are also presented which provide a more accurate treatment of scattering from the penetrable rubber foam. These simulations resolve high-order, very narrow focusing in regions close to the lens (~5 wavelengths). A comparison of spherical and coma aberration between a conventional and FZP lens are also reported. [Work sponsored by the Office of Naval Research.]

9:00

5aUW5. High frequency p-wave attenuation dispersion in water-saturated granular medium with bimodal grain size distribution. Haesang Yang (Underwater Acoust. Lab., Dept. of Naval Archi. & Ocean Eng., Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 151-744, South Korea, coupon3@snu.ac.kr), Keunhwa Lee (Sejong Univ., Seoul, South Korea), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Compressional wave attenuation in water-saturated granular medium depends on both the frequency and grain size. In recent series of high frequency measurements on porous medium, such as water-saturated glass beads and marine sediments, the effect of grain size distribution on attenuation has been investigated, and a variety of grain size dependence on attenuation has been reported. Particularly, dispersion relation in the granular medium with bimodal grain size distribution is significantly different from that in the medium with unimodal grain size distribution. In this study, two sets of glass beads experiments employing unimodal and bimodal grain size distributions are performed. In order to examine wave propagation phenomenon in the bimodal grains, the attenuation dispersion is represented as a function of three parameters; porosity, volume fraction of scatterers, and Rayleigh parameter *kd*. Heuristic dispersion model considering scattering phenomenon will also be discussed for comparison with measurements.

9:15

5aUW6. Equations describing finite-amplitude effects in acoustic fish abundance estimation. Per Lunde (Univ. of Bergen, Allegaten 55, Bergen N-5020 Bergen, Norway, per.lunde@ift.uib.no)

Species identification of fish and abundance estimation of zooplankton employ frequencies typically in the range of 20–400 kHz. Estimates are based on measurement of (a) the single-target backscattering cross section, σ_{bs}, used in echo sounder calibration and fish target strength measurements; and (b) the volume backscattering coefficient, s_v, used in oceanic surveying. The power budget equations used today for σ_{bs} and s_v preassume small-amplitude signals (i.e., linear sound propagation). For operating frequencies about 100 kHz and higher, finite-amplitude (nonlinear) sound propagation effects in seawater may cause measurements errors at accessible echo sounder transmit power levels. In the present work, power budget equations are derived for σ_{bs} and s_v that account for finite-amplitude sound propagation effects. The expressions derived can be used to establish upper limits for echo sounder transmit power levels, in order to reduce finite-amplitude errors in calibration and surveying. They can alternatively be used to develop correction factors for calibration and/or survey data already subjected to finite-amplitude errors. The expressions derived are fully consistent with the small-signal expressions used in fisheries acoustics today.

9:30

5aUW7. A cavitation threshold for transient signals applied to laboratory-scale sparker-induced pulses. J. James Esplin (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jje166@psu.edu), Benjamin Kim (Appl. Res. Lab., Penn State Univ., University Park, PA), and R. Lee Culver (Appl. Res. Lab., Penn State Univ., State College, PA)

The phenomenon known as cavitation can occur when a volume of liquid is subjected to a pressure that falls below a "cavitation threshold". Following this cavitation inception, a rupturing of the fluid or rapid growth of microbubbles occurs. The cavitation threshold is typically thought to be equal to the vapor pressure of the fluid; however, laboratory experiments involving underwater high-amplitude sparker-induced pulses have demonstrated that this is not necessarily the case. This presentation introduces a generalized threshold for transient acoustic pulses based on previous work of a threshold for constant-frequency transient signals. The output of this transient cavitation threshold will be compared against simulation and experiment.

9:45

5aUW8. Nonlinear acoustic pulse propagation in ocean waveguides containing an elastic seabed. Joseph T. Maestas (Appl. Mathematics & Statistics, Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jmaestas@mines.edu) and Jon M. Collis (Lincoln Lab., Lexington, MA)

High intensity underwater sources, such as explosions, generate nonlinear finite-amplitude pulses that behave differently than linear acoustic pulses within shallow water waveguides. The nonlinearity is known to decrease the critical angle for total internal reflection from that of the linear case when the seafloor is approximated as a fluid. However, this result has not been extensively studied for elastic seafloors where shear waves are present. In this work, a time-domain model is developed assuming an isotropic linear elastic bottom, inviscid water column, and allowing for nonlinear advective acceleration and a nonlinear equation of state. The model is numerically implemented using a high-order Godunov scheme and then benchmarked against tank experiment data for the linear case. The nonlinear model is used to study the critical grazing angle for ocean bottoms of varying shear speeds to determine the combined effect of nonlinearity and elasticity on bottom penetration.

10:00–10:15 Break

10:15

5aUW9. Estimation of cylinder orientation using autonomous underwater vehicle mapping of bistatic scattered fields. Erin M. Fischell and Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu)

When an aspect-dependent target is insonified by an acoustic source, distinct features are produced in the resulting bistatic scattered field. These features were exploited in a process for estimating target aspect angles that was demonstrated on a real-world data set using models produced using simulation data. Bistatic scattering data was collected during an experiment in November 2014 in Massachusetts Bay using a ship-based acoustic source producing 7-9kHz LFM chirps and a steel pipe target. The true target orientation was unknown, as the target was dropped from the ship with no rotation control. The Autonomous Underwater Vehicle (AUV) Unicorn, fitted with a 16-element nose array and data acquisition payload, was deployed in broadside data collection behaviors around the target, and the ship was moved to create two target aspects. Scattering data was collected for each target aspect angle. A Support Vector Machine (SVM) regression model was trained using simulated scattering bistatic field data from the OASES-SCATT simulation package. This model was used to estimate the target aspect angles using the real data collected by the AUV during the experiment. The aspect angle estimates were consistent with experimental observations of relative source positioning based on ship position.

10:30

5aUW10. Extending the distributed-basis transition matrix for acoustic target scattering to highly oblate elastic objects in free-field. Raymond Lim (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Code X11, Panama City, FL 32407-7001, raymond.lim@navy.mil)

In previously presented work, a variant of Waterman's transition (T) matrix utilizing an ansatz for problematic outgoing basis functions in standard formulations was proposed and demonstrated to improve the stability of free-field acoustic scattering calculations for elongated axisymmetric elastic objects. The ansatz replaced the basis causing instability with a non-local basis consisting of low-order spherical functions made into a complete set by distributing the functions along the axis within the object. Unfortunately, as pointed out by Doicu, *et al*. [*Acoustic & Electromagnetic Scattering Analysis Using Discrete Sources*, Academic Press, London, 2000], these bases are not as useful for expanding surface fields on oblate axisymmetric shapes. However, they suggested an alternative basis for such shapes in the form of low-order spherical functions made complete by analytically continuing them into a complex plane of the axial coordinate of the object and distributing them along the imaginary axis of this axial coordinate. This presentation will show how this alternative performs in our T-matrix formulation for highly oblate axisymmetric objects, discuss sources of residual noise, and suggest ways to remediate the noise for such shapes. [Work supported by ONR.]

10:45

5aUW11. Scattering from a partially buried target: A modified acoustic ray model. Steven G. Kargl, Aubrey L. Espana, and Kevin L. Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, kargl@uw.edu)

An acoustic ray model, recently presented by Kargl *et al.* (*IEEE J. Ocean. Eng*., DOI: 10.1109/JOE.2014.2356934), describes the scattering of sound from a target within a homogeneous waveguide. For a proud target, four ray paths account for the interaction of incident sound with the target and its local environment. In marine environments, a target at the water-sediment boundary can be partially or completely buried. Ray paths that may reach a target then depend on the burial state. Modifications to the ray model are presented. The simplest approach introduces a Heaviside step function into the model, which abruptly turns a ray contribution off as a ray's point of impact on a target transitions from above the boundary to below. In geometric acoustics, the Heaviside step function is a zeroth order approximation to the Fresnel integral for the diffraction of sound at a shadow boundary into a shadow region. The Fresnel integral provides a smooth transition in the intensity across the boundary. Similarly, a smooth transition should occur within a modified ray model for partially buried targets. The modified ray model will be compared to finite-element model results for a solid aluminum cylinder and an aluminum replica of a underwater unexploded ordnance. [Research supported by SERDP and ONR.]

11:00

5aUW12. Deep ocean vector sensor array performance metrics. Gabriel P. Kniffin and Lisa Zurk (Elec. and Comput. Eng., Portland State Univ., P. O. Box 751 - ECE, Portland, OR 97207, kniffing@pdx.edu)

Recent work in passive sonar has drawn interest in the potential for vertical line arrays (VLAs) deployed below the critical depth—the depth in the deep ocean at which the sound speed below the channel axis reaches the sound speed near the surface. Such arrays can take advantage of propagation via the reliable acoustic path (RAP), which has been shown to improve the signal-to-noise ratio (SNR) of received signals from sources at or near the surface at moderate ranges. The potential of these deep ocean VLAs has spawned further interest in the design of vector sensor VLAs that would allow azimuthal rejection and additional array gain over conventional pressure sensor VLAs. This work will present simulation results that explore and quantify the performance of such vector sensor VLAs deployed in the deep ocean in terms of surface area coverage, detection probability, and operational lifetime. The potential use of these deep ocean vector sensor arrays to estimate source depth using the depth-harmonic interference between direct and surface-reflected acoustic arrivals will also be discussed.

11:15

5aUW13. Space-time block coding for undersea acoustic links. Sabna N, Revathy R, and P. R. S. Pillai (Dept. of Electronics - CUCENTOL Lab, Cochin Univ. of Sci. and Technol., CUCENTOL Lab, Ernakulam, Kerala 682022, India, sabnan@yahoo.com)

Current research in undersea communication scenario focuses on the applications of undersea acoustic networks in environmental data collection, pollution monitoring, offshore surveillance, coastal surveillance, etc. The various constraints encountered in undersea acoustic communications are mainly low bandwidth, high latency, high failure rates of acoustic links, and multipath propagation effects. Multipath effects in undersea environments are due to the reflection of sound by the sea surface and sea bottom as well as refraction of sound in water. As a result, the receiver gets a bewildering mix of signals. The receiving subsystems of the undersea networks perform special signal processing techniques for regenerating the data streams from the multipath composite signals in a MIMO configurable network. The performance of Space-Time Block Coding (STBC) diversity technique has been adopted in this paper for improving the reliability of the undersea network. In the simplest form of MIMO configuration, two transmitting transducers send out the data stream and its modified versions in two time slots on independently fading environments, leading to substantial improvements in the error rate performance. For a shallow water medium range channel, which exhibits the Rayleigh fading characteristics, the error rate performance has been found to be 7×10^{-6} for 16-QAM based STBC.

This document is frequently updated; the current version can be found online at the Internet site: http://scitation.aip.org/content/asa/journal/jasa/info/authors and on JASA's Editorial Manager submission site, www.editorialmanager.com/JASA.

Information for contributors to the Journal of the Acoustical Society of America (JASA)

Editorial Staff[a)]

Journal of the Acoustical Society of America, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300

The procedures for submitting manuscripts to the *Journal of the Acoustical Society of America* are described. The text manuscript, the individual figures, a reviewer PDF, and a cover letter are each uploaded as separate files to the *Journal's* Manuscript Submission and Peer Review System. The required format for the text manuscript is intended so that it will be easily interpreted and copy-edited during the production editing process. Various detailed policies and rules that will produce the desired format are described, and a general guide to the preferred style for the writing of papers for the *Journal* is given. Criteria used by the editors in deciding whether or not a given paper should be published are summarized.

TABLE OF CONTENTS

[a)]E-mail: jasa@acousticalsociety.org

I. INTRODUCTION

The present document is intended to serve jointly as (i) a set of directions that authors should follow when submitting articles to the *Journal of the Acoustical Society of America* and as (ii) a style manual that describes those stylistic features that are desired for the submitted manuscript. Authors may refer to recent issues of the *Journal* for examples of how specific style issues are handled.

II. ONLINE HANDLING OF MANUSCRIPTS

All new manuscripts intended for possible publication in the *Journal of the Acoustical Society of America* should be submitted online. The steps involved in the processing of manuscripts that lead from the initial submission through the peer review process to the transmittal of an accepted manuscript to the production editing office are handled by a computerized system referred to here as the Editorial Manager (EM) system. The Acoustical Society of America contracts with Aries Systems, Inc. for the use of this system. There is one implementation that is used for most of the material that is submitted to the *Journal of the Acoustical Society of America* (JASA) and a separate implementation for the special section *JASA Express Letters (JASA-EL)* of the *Journal.*

A. Registration

Everyone involved in the handling of manuscripts in the *Journal's* editorial process must first register with the *Journal's* implementation of the EM system, and the undertaking of separate actions, such as the submission of a manuscript, requires that one first log-in to the system at www.editorialmanager.com/JASA or www.editorialmanager.com/JASA-EL.

If you have never logged into the system, you will need to get a user name and password. Many ASA members are already in the database, so if you are a member, you in principle may already have a user name and password.

Once you have your "user name" and "password" you go to the log-in page, and give this information when you log-in.

B. Overview of the editorial process

(1) An author denoted as the corresponding author submits a manuscript for publication in the *Journal.*
(2) A quick quality control check is done on the manuscript. If there are too many (n>15) errors in the submitted manuscript, it will be returned to the corresponding author to fix them and resubmit.
(3) One of the *Journal's* Associate Editors is recruited to handle the peer-review process for the manuscript.
(4) The Associate Editor recruits reviewers for the manuscript via the online system.
(5) The reviewers critique the manuscript, and submit their comments online via the EM system.
(6) The Associate Editor makes a decision regarding the manuscript, and then composes online an appropriate decision letter, which may include segments of the reviews, and which may include attachments.
(7) The *Journal's* staff transmits a letter composed by the Associate Editor to the corresponding author. This letter describes the decision and further actions that can be taken.

If revisions to the manuscript are invited, the author may resubmit a revised manuscript, and the process cycle is repeated. To submit a revision authors should use the link provided in the decision message.

C. Preparation for online submission

Before one begins the process of submitting a manuscript online, one should first read the document *Ethical Principles of the Acoustical Society of America for Research Involving Human and Non-Human Animals in Research and Publishing and Presentations* which is reached from the site <http://scitation.aip.org/content/asa/journal/jasa/info/authors>. During the submission, you will be asked if your research conformed to the stated ethical principles and if your submission of the manuscript is in accord with the ethical principles that the Acoustical Society has set for its journals. If you cannot confirm that your manuscript and the research reported are in accord with these principles, then you should not submit your manuscript.

Another document that you should first read is the document *Transfer of Copyright Agreement,* which is downloadable from the same site. When you submit your manuscript online you will be asked to certify that you and your co-authors agree to the terms set forth in that document. What is in that document has been carefully worded with extensive legal advice and which has been arrived at after extensive discussion within the various relevant administrative committees of the Acoustical Society of America. It is regarded as a very liberal document in terms of the rights that are allowed to the authors. One should also note the clause: The author(s) agree that, insofar as they are permitted to transfer copyright, all copies of the article or abstract shall include a copyright notice in the ASA's name. (The word "permitted" means permitted by law at the time of the submission.) The terms of the copyright agreement are non-negotiable. The Acoustical Society does not have the resources or legal assistance to negotiate for exceptions for individual papers, so please do not ask for such special considerations. Please read the document carefully and decide whether you can provide an electronic signature (clicking on an appropriate check box) to this agreement. If you do not believe that you can in good conscience give such an electronic signature, then you should not submit your manuscript.

Given that one has met the ethical criteria and agreed to the terms of the copyright transfer agreement, and that one has decided to submit a manuscript, one should first gather together the various items of information that will be requested during the process, and also gather together various files that one will have to upload.

Information that will be entered into the EM submission form and files to be uploaded include:

(1) Data for each of the authors:
    (a) First name, middle initial, and last name
    (b) E-mail address

(c) Work telephone number
(d) Postal address (required for corresponding author, otherwise optional)

(2) Title and running title of the paper. The running title is used as part of the table of contents on the journal cover. (The title is preferably limited to 17 words and the running title is limited to six words and up to 50 characters and spaces; neither may include non-obvious acronyms or any words explicitly touting novelty.)
(3) Abstract of the paper. (This must be in the form of a single paragraph and is limited to 200 words for regular articles and to 100 words for Letters to the Editor. (Authors would ordinarily do an electronic pasting from a text file of their manuscript.)
(4) Four keywords that characterize the subject matter of the paper.
(5) A short prioritized list of Associate Editors suggested for the handling of the manuscript. (EM currently limits to one, but that will be expanded.)
(6) Contact information (name, e-mail address, and institution) of suggested reviewers (if any), and/or names of reviewers to exclude.
(7) Cover letter file. This should supply additional information that should be brought to the attention of the editor(s) and/or reviewer(s).
(8) Properly prepared manuscript/article file in LaTeX or Word format. (The requirements for a properly prepared manuscript are given further below.) It must be a single stand-alone file. If the author wishes to submit a LaTeX file, the references should be included in the file, not in a separate BibTeX file. Authors should take care to ensure that the submitted manuscript/article file is of reasonable length.
(9) Properly prepared figure files in TIFF, PS, JPEG, or EPS (see also Section V. H); one file for each cited figure number. The uploading of figures in PDF format is not allowed. (The captions should be omitted, and these should appear as a list in the manuscript itself.) The figures should not have the figure numbers included on the figures in the files. Authors may upload figures in a zip file. For figures without subparts (as well as for figures having subparts built in to a single file), the uploaded file should be named "Figure1.nnn" where "nnn" is the type of graphic file (.jpg, .eps, etc.). For compound figures uploaded as separate files, the individual files should be named Figure1a.nnn, Figure1b.nnn, etc., where "nnn" is the correct filetype/file extension. In order to maintain online color as a free service to authors, the *Journal* cannot accept multiple versions of the same file. Authors may not submit two versions of the same illustration (e.g., one for color and one for black & white). When preparing illustrations that will appear in color in the online *Journal* and in black & white in the printed *Journal,* authors must ensure that: (i) colors chosen will reproduce well when printed in black & white and (ii) descriptions of figures in text and captions will be sufficiently clear for both print and online versions. For example, captions should contain the statement "(Color online)." If one desires color in both versions, these considerations are irrelevant, although the authors must guarantee that mandatory additional publication charges will be paid.
(10) Supplemental files (if any) that might help the reviewers in making their reviews. If the reading of the paper requires prior reading of another paper that has been accepted for publication, but has not yet appeared in print, then a PDF file for that manuscript should be included as a supplementary file. Also, if the work draws heavily on previously published material which, while available to the general public, would be time-consuming or possibly expensive for the reviewers to obtain, then PDF files of such relevant material should be included.
(11) Archival supplemental materials to be published with the manuscript in AIP Publishing's Supplemental Materials electronic depository.

In regard to the decision as to what formats one should use for the manuscript and the figures, a principal consideration may be that the likelihood of the published manuscript being more nearly to one's satisfaction is considerably increased if AIP Publishing, during the production process, can make full or partial use of the files you submit. There are conversion programs, for example, that will convert LaTeX and MS Word files to the typesetting system that AIP Publishing uses. If your manuscript is not in either of these formats, then it will be completely retyped. If the figures are submitted in EPS, PS, JPEG, or TIFF format, then they will probably be used directly, at least in part. The uploading of figures in PDF format is not allowed.

D. Steps in online submission

After logging in, one is brought to the EM author main page, and can select the option of submitting a new manuscript. The resulting process leads the corresponding author through a sequence of screens.

The submit screen will display a series of fairly self-explanatory tabs. Clicking on these tabs displays the tasks that must be completed for each step in the submission.

After submission, all of the individual files, text and tables, plus figures, that make up the full paper will be merged into a single PDF file. One reason for having such a file is that it will generally require less computer memory space. Another is that files in this format are easily read with any computer system. However, the originally submitted set of files, given the acceptance for publication, will be what is submitted to the Production Editing office for final processing.

E. Quality check by editorial office

Upon receiving system notification of a submission, staff members in the Editorial Office check that the overall submission is complete and that the files are properly prepared and suitable for making them available to the Associate Editors and the reviewers. If all is in order, the Manuscript Coordinator initiates the process, using the keywords and suggested Associate Editor list supplied by the author, to recruit an Associate Editor who is willing to handle the manuscript. At this time the author also receives a "confirmation of receipt" e-mail message. If the staff members deem that there are numerous or serious submission defects that should be addressed, then the author receives a "quality check" e-mail message. If there are only a small number of defects, the e-mail message may give an explicit description of what is needed. In some cases, when the defects are very numerous, and it is apparent that the author(s) are not aware that the *Journal* has a set of format requirements, the e-mail message may simply ask the authors to read the instructions (i.e., the present document) and to make a reasonable attempt to follow them.

III. PUBLICATION CHARGES

A. Mandatory charges

Papers of longer length or with color figures desired for the print version of the *Journal* will not be published unless it is first agreed that certain charges will be paid. If it is evident that there is a strong chance that a paper's published length will exceed 12 pages, the paper will not be processed unless the authors guarantee that the charges will be paid. If the paper's published length exceeds 12 pages or more, there is a mandatory charge of $80 per page for the entire article. (The mandatory charge for a 13 page article, for example, would be $1040, although there would be no mandatory charge if the length were 12 pages.)

Manuscripts should not exceed 10,500 words [approximately twelve (12) printed journal pages]. Abstract, title, author list, references, and acknowledgments are all excluded from the 10,500-word limit. Figures, tables, and equations, however, are included and must be accounted for by calculating a word count equivalent to the space they occupy. Circumvention of the length limitation is contrary to the purpose of this journal.

Please use these guidelines for estimating length.

TeX users

Authors are advised to use the article class of TeX. If the version of the manuscript obtained using the "reprint" option fits on twelve (12) pages with a font size of 12 points, the length should be acceptable.

Word users

Highlight the manuscript text, excluding abstract, author list, acknowledgments and references, and note the word count at the bottom of the screen. Add to that the word-count-equivalents for figures, tables, and equations as follows:

Figures: An average single-column figure will displace 220 words. For a more accurate estimation, use the following: 150/aspect ratio + 20 words for single-column figures and 300/0.5 x aspect ratio + 40 words for double-column figures. Aspect ratio = width/height.

Tables: 6.5 words per line, plus 13 words for single-column tables. 13 words per line, plus 26 words for double-column tables.

Equations: 16 words per row for single-column equations. 32 words per row for double-column equations.

If the total number of words (text + figures + tables + equations) is 10,500 or less, the length is acceptable.

Color figures can be included in the online version of the *Journal* with no extra charge, providing that these appear suitably as black and white figures in the print version.

The charges for inclusion of color figures in the print version of the *Journal* are $325 per figure file.

If an author's institution or research sponsor is unwilling to pay such charges, the author should make sure that all of the figures in the paper are suitable for black and white printing, and that the estimated length is manifestly such that it will not lead to a printed paper that exceeds 12 pages.

JASA now offers a "gold" open access option, the price of which is $2200 USD. If an open access paper runs beyond 12 pages, overpage fees are only due on the pages beyond 12.

B. Optional charges

To encourage a large circulation of the *Journal* and to allow the inclusion of a large number of selected research articles within its volumes, the *Journal* seeks partial subsidization from the authors and their institutions. Ordinarily, it is the institutions and/or the sponsors of the research that undertake the subsidization. Individual authors must ask their institutions or whatever agencies sponsor their research to pay a page charge of $80 per printed page to help defray the publication costs of the *Journal.* (This is roughly 1/3 of the actual cost per page for the publication of the *Journal.*) The institutions and the sponsoring agencies have the option of declining, although a large fraction of those asked do pay them. The review and selection of manuscripts for publication proceeds without any consideration on the part of the Associate Editors as to whether such page charges will be honored. The publication decision results after consideration of the factors associated with peer review; the acceptance of the page charges is irrelevant.

C. Waived charges

To encourage submission of review papers, tutorial papers, forum papers, and special external (to our specialties) papers, all of which are invited, we now waive the publication fee for these article types. However, a fee for (optional) color in the print version will be requested for such articles.

D. Payment of publication charges—RightsLink

When your page proofs are ready for your review, you will receive an e-mail from AIP Publishing Production Services. It will include a link to an online RightsLink site where you can pay your voluntary or mandatory page charges, color figure charges, or to order reprints of your article. If you are unable to remit payment online, you will find instructions for requesting a printed invoice so that you may pay by check or wire transfer.

IV. FORMAT REQUIREMENTS FOR MANUSCRIPTS

A. Overview

For a manuscript to pass the initial quality control, it is essential that it adhere to a general set of formatting requirements. Such vary from journal to journal, so one should not assume that a manuscript appropriate for another journal's requirements would be satisfactory for the *Journal* *of the Acoustical Society of America.* The reasons for the *Journal's* requirements are partly to ensure a uniform style for publications in the *Journal* and partly to ensure that the copy-editing process will be maximally effective in producing a quality publication. While some submitted papers will need very few or no corrections, there is a sufficiently large number of accepted papers of high technical merit that need such editing to make it desirable that all submissions are in a format that amply allows for this.

The following is a list of some of the more important requirements. (More detailed requirements are given in the sections that follow.)

(1) The manuscript must be paginated, starting with the first page.
(2) The entire manuscript must be doubled-spaced. This includes the author addresses, the abstract, the references, and the list of figure captions. It should contain no highlighting.
(3) The title and author list is on the first page. The abstract is ordinarily on a separate page (the second page) unless there is sufficient room on the title page for it, within the constrains of ample margins, 12 pt type, double-spacing, and ample white space. The introduction begins on a separate page following the page that contains the abstract.
(4) The title must be in lower case, with the only capitalized words being the first word and proper nouns.
(5) No acronyms should be in the title or the running title unless they are so common that they can be found in standard dictionaries.
(6) No unsupported claims for novelty or significance should appear in the title or abstract, such as the use of the words *new, original, novel, important,* and *significant.*
(7) The abstract should be one paragraph and should be limited to 200 words (100 words for Letters to the Editor).
(8) Major section headings should be numbered by capital roman numerals, starting with the Introduction. Text of such headings should be in capital letters.
(9) Reference citations should include the full article titles and page ranges of all cited papers.
(10) There should be no personal pronouns in the abstract.
(11) No more than one-half of the references should be to the authors themselves.
(12) The total number of figures should not ordinarily be more than 20 (see Section V. H).

B. PDF for reviewers

A PDF file with line numbers and inline figures and captions needs to be provided by the author(s) for each manuscript, for the ease of the reviewers.

C. Keyboarding instructions

Each submitted paper, even though submitted online, should correspond to a hard copy manuscript. The electronic version has to be prepared so that whatever is printed-out will correspond to the following specifications:

(1) The print-out must be single sided.
(2) The print-out must be configured for standard US letter paper (8.5″ by 11″).
(3) The text on any given page should be confined to an area not to exceed 6.5″ by 9″. (One inch equals 2.54 cm.) All of the margins when printed on standard US letter paper should be at least 1″.
(4) The type font must be 12 pt, and the line spacing must correspond to double spacing (approximately 1/3″ or 0.762 cm per line of print). The fonts used for the text must be of a commonly used easily readable variety such as Times, Helvetica, New York, Courier, Palatino, and Computer Modern.
(5) The authors are requested to use computers with adequate word-processing software in preparing their manuscripts. Ideally, the software must be sufficiently complete that all special symbols used in the manuscript are printed. (The list of symbols available to AIP Publishing for the publication of manuscripts includes virtually all symbols that one can find in modern scientific literature. Authors should refrain from inventing their own symbols.) It is preferred that vectors be designated by boldface symbols within a published paper rather than by arrows over the symbols.
(6) Manuscript pages must be numbered consecutively, with the title page being page 1.

D. Order of pages

The manuscript pages must appear in the following order:

(1) Title page. (This includes the title, the list of authors, their affiliations, an abbreviated title for use as a running title in the published version, and any appropriate footlines to title or authors.)
(2) Abstract page, which may possibly be merged with the title page if there is sufficient room. Please note that the *Journal* requires the abstract to be typed double spaced, just as for all of the remainder of the manuscript.
(3) Text of the article. This must start on a new page.
(4) Acknowledgments.
(5) Appendixes (if any).
(6) Textual footnotes. (Allowed only if the paper cites references by author name and year of publication.)
(7) References. (If the paper cites references by labeling them with numbers according to the order in which they appear, this section will also include textual footnotes.)
(8) Tables, each on a separate page and each with a caption that is placed above the table.
(9) Collected figure captions.

Figures should not be included in the “Article” file.

E. Title page of manuscript

The title page should include on separate lines, with appropriate intervening spacing: The article title, the name(s) of author(s), one complete affiliation for each author, and the date on which the manuscript is uploaded to the JASA manuscript submission system.

With a distinctive space intervening, the authors must give, on a separate line, a suggested running title of six words or less that contains a maximum of 50 characters. The running title appears on the front cover of the *Journal* as part of an abbreviated table of contents, and it is important that it give a nontrivial indication of the article's content, although some vagueness is to be expected.

Titles should briefly convey the general subject matter of the paper and should not serve as abstracts. The upper limit is set at 17 words. They must be written using only words and terminology that can be found in standard unabridged US English dictionaries or in standard scientific/technical dictionaries, and they must contain no acronyms other than those that can be found in such dictionaries. Ideally, titles should be such that they contain appropriate keywords. This will enable a reader doing a computer-assisted search to determine whether the paper has any relevance to a given research topic. Begin the first word of the title with a capital letter; thereafter capitalize only proper nouns. The *Journal* discourages the use of subjective words such as "original," "new," "novel," "important," and "significant" in the title. In general, words whose sole purpose is to tout the importance of a work are regarded as unnecessary; words that clarify the nature of the accomplishment are preferred.

In the list of authors, to simplify later indexing, adopt one form of each name to use on the title pages of all submissions to the *Journal.* It is preferred that the first name be spelled out, especially if the last name is a commonly encountered last name. If an author normally uses the middle name instead of the first name, then an appropriate construction would be one such as J. John Doe. Names must be written with last name (family name) given last. Omit titles such as Professor, Doctor, Colonel, Ph.D., and so on.

Each author may include only one current affiliation in the manuscript. Put the author's name above the institutional affiliation. When there is more than one author with the same institutional affiliation, put all such names above the stating of that affiliation. (See recent issues of the *Journal* for examples.)

In the stating of affiliations, give sufficient (but as briefly as possible) information so that the corresponding author may be contacted by mail and/or e-mail by interested readers. Do not give websites, telephone numbers, or FAX numbers. Names of states and countries should be written out in full. If a post office box should be indicated, append this to the zip code (as in 02537-0339). Use no abbreviations other than DC (for District of Columbia). If the address is in the United States, include "USA."

The preferred order of listing of authors is in accord with the extent of their contributions to the research and to the actual preparation of the manuscript. (Thus, the last listed author is presumed to be the person who has done the least.)

The stated affiliation of any given author should be that of the institution that employed the author at the time the work was done. In the event an author was employed simultaneously by several institutions, the stated affiliation should be that through which the financial support for the research was channeled. If the current (at the time of publication) affiliation is different, then that should be stated in a footline. If an author is deceased then that should be stated in a footline. (Footlines are discussed further below.)

There is no upper limit to the number of authors of any given paper. If the number becomes so large that the appearance of the paper when in print could look excessively awkward, the authors will be given the option of not explicitly printing the author affiliations in the heading of the paper. Instead, these can be handled by use of footlines as described below. The *Journal* does not want organizations or institutions to be listed as authors. If there are a very large number of authors, those who made lesser contributions can be designated by a group name, such a name ending with the word "group." A listing of the members of the group possibly including their addresses should be given in a footline.

Footlines to the title and to the authors' names are consecutively ordered and flagged by lowercase alphabetical letters, as in Fletcher[a)], Hunt[b)], and Lindsay[c)]. If there is any history of the work's being presented or published in part earlier, then a footline flag should appear at the end of the title, and the first footline should be of the form exemplified below:[1]

[a)]Portions of this work were presented in "A modal distribution study of violin vibrato," Proceedings of International Computer Music Conference, Thessaloniki, Greece, September 1997, and "Modal distribution analysis of vibrato in musical signals," Proceedings of SPIE International Symposium on Optical Science and Technology, San Diego, CA, July 1998.

Authors have the option of giving a footline stating the e-mail address of one author only (usually the corresponding author), with an appropriate footline flag after that name and with each footline having the form:

[b)]Electronic mail: name@servername.com

F. Abstract page

Abstracts are often published separately from actual articles, and thus are more accessible than the articles themselves to many readers. Authors consequently must write abstracts so that readers without immediate access to the entire article can decide whether the article is worth obtaining. The abstract is customarily written last; the choice of what should be said depends critically on what is said in the body of the paper itself.

The abstract should not be a summary of the paper. Instead, it should give an accurate statement of the subject of the paper, and it should be written so that it is intelligible to a broad category of readers. Explicit results need not be stated, but the nature of the results obtained should be stated. Bear in mind that the abstract of a journal article, unlike the abstract of a talk for a meeting, is backed-up by a written article that is readily (if not immediately) accessible to the reader.

Limit abstracts to 200 words (100 words for Letters to the Editor). Do not use footnotes. If the authors decide that it is imperative to cite a prior publication in the abstract, then the reference should be embedded within the text and enclosed within square brackets. These should be in one of the two standard JASA formats discussed further below, but titles of articles need not be given. The abstract should contain no acknowledgments. In some circumstances, abstracts of longer than 200 words will be allowed. If an author believes that a longer abstract is essential for the paper, they should send an e-mail message to jasa@aip.org with the subject line "Longer abstract requested." The text of the desired abstract should be included in the memo, along with a statement of why the author believes the longer abstract is essential. The abstract will be reviewed by the editors, and possibly a revised wording may be suggested.

Personal pronouns and explicit claims as to novelty should be assiduously avoided. Do not repeat the title in the abstract, and write the abstract with the recognition that the reader has already read the title. Avoid use of acronyms and unfamiliar abbreviations. If the initial writing leads to the multiple use of a single lengthy phrase, avoid using an author-created acronym to achieve a reduction in length of the abstract. Instead, use impersonal pronouns such as *it* and *these* and shorter terms to allude to that phrase. The shortness of the abstract reduces the possibility that the reader will misinterpret the allusion.

G. Section headings

The text of a manuscript, except for very short Letters to the Editor, is customarily broken up into sections. Four types of section headings are available: principal heading, first subheading, second subheading, and third subheading. The principal headings are typed boldface in all capital letters and appear on separate lines from the text. These are labeled by uppercase roman numerals (I, II, III, IV, etc.), with the introductory section being principal section I. First subheadings are also typed on separate lines; these are labeled by capital letters: A, B, C, etc. The typing of first subheadings is boldface, with only the first word and proper nouns being capitalized. Second subheadings are ordered by numbers (1, 2, 3, etc.) and are also typed on separate lines. The typing of second subheadings is italic boldface, also with only the first word and proper nouns capitalized. Third subheadings appear in the text at the beginning of paragraphs. These are labeled by lowercase letters (a, b, c, etc.) and these are typed in italics (not boldfaced). Examples of these types of headings can be found in recent issues of the *Journal.* (In earlier issues, the introduction section was not numbered; it is now required to be numbered as the first principal section.)

Headings to appendixes have the same form as principal headings, but are labeled by uppercase letters, with an optional brief title following the identification of the section as an appendix, as exemplified below:

APPENDIX C: CALCULATION OF IMPEDANCES

If there is only one appendix, the letter designation should be omitted.

V. STYLE REQUIREMENTS

A. Citations and footnotes

Regarding the format of citations made within the text, authors have two options: (1) textual footnote style and (2) alphabetical bibliographic list style.

In the *textual footnote style,* references and footnotes are cited in the text by superscripted numerals, as in "the basic equation was first derived by Rayleigh[44] and was subsequently modified by Plesset.[45]" References and footnotes to text material are intercalated and numbered consecutively in order of first appearance. If a given reference must be cited at different places in the text, and the citation is identical in all details, then one must use the original number in the second citation.

In the *alphabetical bibliographic list style,* footnotes as such are handled as described above and are intended only to explain or amplify remarks made in the text. Citations to specific papers are flagged by parentheses that enclose either the year of publication or the author's name followed by the year of publication, as in the phrases "some good theories exist (Rayleigh, 1904)" and "a theory was advanced by Rayleigh (1904)." In most of the papers where this style is elected there are no footnotes, and only a bibliographic list ordered alphabetically by the last name of the first author appears at the end of the paper. In a few cases,[2] there is a list of footnotes followed by an alphabetized reference list. Within a footnote, one has the option of referring to any given reference in the same manner as is done in the text proper.

Both styles are in common use in other journals, although the *Journal of the Acoustical Society of America* is one of the few that allows authors a choice. Typically, the textual footnote style is preferred for articles with a smaller number of references, while the alphabetical bibliographic list style is preferred for articles with a large number of references. The diversity of the articles published in the *Journal* makes it infeasible to require just one style unilaterally.

B. General requirements for references

Regardless of what reference style the manuscript uses, the format of the references must include the titles of articles. For articles written in a language other than English, and for which the Latin alphabet is used, give the actual title first in the form in which it appeared in the original reference, followed by the English translation enclosed within parentheses. For titles in other languages, give only the English translation, followed by a statement enclosed in parentheses identifying the language of publication. Do not give Latin-alphabet transliterations of the original title. For titles in English and for English translations of titles, use the same format as specified above for the typing of the title on the title page. Begin the first word of the title with a capital letter; thereafter capitalize only those words that are specified by standard dictionaries to be capitalized in ordinary prose.

One must include only references that can be obtained by the reader. One should also not cite any paper that has only been submitted to a journal; if it has been accepted, then the citation should include an estimated publication date. If one cites a reference, then the listing must contain enough information that the reader can obtain the paper. If theses, reports, or proceedings are cited, then the listing must contain specific addresses to which one can write to buy or borrow the reference. In general, write the paper in such a manner that its understanding does not depend on the reader having access to references that are not easily obtained.

Authors should avoid giving references to material that is posted on the internet, unless the material is truly archival, as is the case for most online journals. If referring to nonarchival material posted on the internet is necessary to give proper credit for priority, the authors should give the date at which they last viewed the material online. If authors have supplementary material that would be of interest to the readers of the article, then a proper posting of this in an archival form is to make use of the AIP Publishing's supplemental material electronic depository. Instructions for how one posts material can be found at the site <http://scitation.aip.org/content/asa/journal/jasa/info/authors>. Appropriate items for deposit include color figures, data tables, and text (e.g., appendixes) that are too lengthy or of too limited interest for inclusion in the printed journal. If authors desire to make reference to materials posted by persons other than by the authors, and if the posting is transitory, the authors should first seek to find alternate references of a more archival form that they might cite instead. In all cases, the reading of any material posted at a transitory site must not be a prerequisite to the understanding of the material in the paper itself, and when such material is cited, the authors must take care to point out that the material will not necessarily be obtainable by future readers.

In the event that a reference may be found in several places, as in the print version and the online version of a journal, refer first to the version that is most apt to be archived.

In citing articles, give both the first and last pages that include it. Including the last page will give the reader some indication of the magnitude of the article. The copying en toto of a lengthy article, for example, may be too costly for the reader's current purposes, especially if the chief objective is merely to obtain a better indication of the actual subject matter of the paper than is provided by the title.

The use of the expression *"et al."* in listing authors' names is encouraged in the body of the paper, but must not be used in the actual listing of references, as reference lists in papers are the primary sources of large data bases that persons use, among other purposes, to search by author. This rule applies regardless of the number of authors of the cited paper.

References to unpublished material in the standard format of other references must be avoided. Instead, append a graceful footnote or embed within the text a statement that you are making use of some material that you have acquired from another person—whatever material you actually use of this nature must be peripheral to the development of the principal train of thought of the paper. A critical reader will not accept its validity without at least seeing something in print. If the material is, for example, an unpublished derivation, and if the derivation is important to the substance of the present paper, then repeat the derivation in the manuscript with the original author's permission, possibly including that person as a co-author.

Journal titles must ordinarily be abbreviated, and each abbreviation must be in a "standard" form. For determination of what abbreviations to use, one can skim the reference lists that appear at the ends of recent articles in the *Journal.* The general style for making such abbreviations (e.g., Journal is always abbreviated by "J.," Applied is always abbreviated by "Appl.," International is always abbreviated by "Int.," etc.) must in any event emerge from a study of such lists, so the authors should be able to make a good guess as to the standard form. Should the guess be in error, this will often be corrected in the copy-editing process. Egregious errors are often made when the author lifts a citation from another source without actually looking up the original source. An author might be tempted, for example, to abbreviate a journal title as "Pogg. Ann.," taking this from some citation in a 19th century work. The journal cited is *Annalen der Physik,* sometimes published with the title *Annalen der Physik und Chemie,* with the standard abbreviation being "Ann. Phys. (Leipzig)." The fact that J. C. Poggendorff was at one time the editor of this journal gives very little help in the present era in distinguishing it among the astronomical number of journals that have been published. For Poggendorff's contemporaries, however, "Pogg. Ann." had a distinct meaning.

Include in references the names of publishers of book and standards and their locations. References to books and proceedings must include chapter numbers and/or page ranges.

C. Examples of reference formats

The number of possible nuances in the references that one may desire to cite is very large, and the present document cannot address all of them; a study of the reference lists at the ends of articles in recent issues in the *Journal* will resolve most questions. The following two lists, one for each of the styles mentioned above, give some representative examples for the more commonly encountered types of references. If the authors do not find a definitive applicable format in the examples below or in those they see in scanning past issues, then it is suggested that they make their best effort to create an applicable format that is consistent with the examples that they have seen, following the general principles that the information must be sufficiently complete that: (1) any present or future reader can decide whether the work is worth looking at in more detail; (2) such a reader, without great effort, can look at, borrow, photocopy, or buy a copy of the material; and (3) a citation search, based on the title, an author name, a journal name, or a publication category, will result in the present paper being matched with the cited reference.

1. Textual footnote style

[1]Y. Kawai, "Prediction of noise propagation from a depressed road by using boundary integral equations," J. Acoust. Soc. Jpn. **56**, 143–147 (2000) (in Japanese).

[2]L. S. Eisenberg, R. V. Shannon, A. S. Martinez, J. Wygonski, and A. Boothroyd, "Speech recognition with reduced spectral cues as a function of age," J. Acoust. Soc. Am. **107**, 2704–2710 (2000).

[3]J. B. Pierrehumbert, *The Phonology and Phonetics of English Intonation* (Ph.D. dissertation, Mass. Inst. Tech., Cambridge, MA, 1980); as cited by D. R. Ladd, I. Mennen, and A. Schepman, J. Acoust. Soc. Am. **107**, 2685–2696 (2000).

[4]F. A. McKiel, Jr., "Method and apparatus or sibilant classification in a speech recognition system," U.S. Patent No. 5,897,614 (27 April 1999). A brief review by D. L. Rice appears in: J. Acoust. Soc. Am. **107**, 2323 (2000).

[5]A. N. Norris, "Finite-amplitude waves in solids," in *Nonlinear Acoustics,*

edited by M. F. Hamilton and D. T. Blackstock (Academic Press, San Diego, 1998), Chap. 9, pp. 263–277.

[6]V. V. Muzychenko and S. A. Rybak, "Amplitude of resonance sound scattering by a finite cylindrical shell in a fluid," Akust. Zh. **32**, 129–131 (1986) [Sov. Phys. Acoust. **32**, 79–80 (1986)].

[7]M. Stremel and T. Carolus, "Experimental determination of the fluctuating pressure on a rotating fan blade," on the CD-ROM: *Berlin, March 14–19, Collected Papers, 137th Meeting of the Acoustical Society of America and the 2nd Convention of the European Acoustics Association* (ISBN 3-9804458-5-1, available from Deutsche Gesellschaft fuer Akustik, Fachbereich Physik, Universitaet Oldenburg, 26111 Oldenburg, Germany), paper 1PNSB_7.

[8]ANSI S12.60-2002 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Institute, New York, 2002).

2. *Alphabetical bibliographic list style*

American National Standards Institute (**2002**). ANSI S12.60 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Institute, New York).

Ando, Y. (**1982**). "Calculation of subjective preference in concert halls," J. Acoust. Soc. Am. **71**(Suppl. 1), S4–S5.

Bacon, S. P. (**2000**). "Hot topics in psychological and physiological acoustics: Compression," J. Acoust. Soc. Am. **107,** 2864(A).

Flatté, S. M., Dashen, R., Munk, W. H., Watson, K. M., and Zachariasen, F. (**1979**). *Sound Transmission through a Fluctuating Ocean* (Cambridge University Press, London), pp. 31–47.

Hamilton, W. R. (**1837**). "Third supplement to an essay on the theory of systems of waves," Trans. Roy. Irish Soc. **17**(Pt. 1), 1–144; reprinted in: *The Mathematical Papers of Sir William Rowan Hamilton, Vol. II: Dynamics,* edited by A. W. Conway and A. J. McConnell (Cambridge University Press, London), pp. 162–211.

Helmholtz, H. (**1859**). "Theorie der Luftschwingungen in Röhren mit offenen Enden" ("Theory of air oscillations in tubes with open ends"), J. reine ang. Math. **57,** 1–72.

Kim, H.-S., Hong, J.-S., Sohn, D.-G., and Oh., J.-E. (**1999**). "Development of an active muffler system for reducing exhaust noise and flow restriction in a heavy vehicle," Noise Control Eng. J. **47,** 57–63.

Simpson, H. J., and Houston, B. H. (**2000**). "Synthetic array measurements for waves propagating into a water-saturated sandy bottom for a smoothed and roughened surface," J. Acoust. Soc. Am. **107,** 2329–2337.

van Bergeijk, W. A., Pierce, J. R., and David, E. E., Jr. (**1960**)**.** *Waves and the Ear* (Doubleday, Garden City, NY), Chap. 5, pp. 104–143.

Other examples may be found in the reference lists of papers recently published in the *Journal.*

D. Figure captions

The illustrations in the *Journal* have *figure captions* rather than *figure titles.* Clarity, rather than brevity, is desired, so captions can extend over several lines. Ideally, a caption must be worded so that a casual reader, on skimming an article, can obtain some indication as to what an illustration is depicting, without actually reading the text of the article. If an illustration is taken from another source, then the caption must acknowledge and cite that source. Various examples of captions can be found in the articles that appear in recent issues of the *Journal.*

If the figure will appear in black and white in the printed edition and in color online, the statement "(Color online)" should be added to the figure caption. For color figures that will appear in black and white in the printed edition of the *Journal,* the reference to colors in the figure may not be included in the caption, e.g., red circles, blue lines.

E. Acknowledgments

The section giving acknowledgments must not be numbered and must appear following the concluding section. It is preferred that acknowledgments be limited to those who helped with the research and with its formulation and to agencies and institutions that provided financial support. Administrators, administrative assistants, associate editors, and persons who assisted in the nontechnical aspects of the manuscript preparation may also be acknowledged. In many cases, sponsoring agencies require that articles give an acknowledgment and specify the format in which the acknowledgment must be stated—doing so is fully acceptable. Generally, the *Journal* expects that the page charges will be honored for any paper that carries an acknowledgment to a sponsoring organization.

F. Mathematical equations

Authors are expected to use computers with appropriate software to typeset mathematical equations.

Authors are also urged to take the nature of the actual layout of the journal pages into account when writing mathematical equations. A line in a column of text is typically 60 characters, but mathematical equations are often longer. To ensure that their papers look attractive when printed, authors must seek to write sequences of equations, each of which fits into a single column, some of which define symbols appearing in another equation, even if such results in a greater number of equations. If an equation whose length will exceed that of a single column is unavoidable, then the authors must write the equation so that it is neatly breakable into distinct segments, each of which fits into a single column. The casting of equations in a manner that requires the typesetting to revert to a single column per page (rather than two columns per page) format must be assiduously avoided. To make sure that this possibility will not occur, authors familiar with desk-top publishing software and techniques may find it convenient to temporarily recast manuscripts into a form where the column width corresponds to 60 text characters, so as to see whether none of the line breaks within equations will be awkward.

Equations are numbered consecutively in the text in the order in which they appear, the number designation is in parentheses and on the right side of the page. The numbering of the equations is independent of the section in which they appear for the main body of the text. However, for each appendix, a fresh numbering begins, so that the equations in Appendix B are labeled (B1), (B2), etc. If there is only one appendix, it is treated as if it were Appendix A in the numbering of equations.

G. Phonetic symbols

The phonetic symbols included in a JASA manuscript should be taken from the International Phonetic Alphabet (IPA), which is maintained by the International Phonetic Association, whose home page is https://www.internationalphoneticassociation.org. The display of the most recent version of the alphabet can be found at https://www.internationalphoneticassociation.org/content/full-ipa-chart.

The total set of phonetic symbols that can be used by AIP Publishing during the typesetting process is the set included among the Unicode characters. This includes most of the symbols and diacritics of the IPA chart, plus a few compiled combinations, additional tonal representations, and separated diacritics. A list of all such symbols is given in the file *phonsymbol.pdf* which can be downloaded by going to the JASA website http://scitation.aip.org/content/asa/journal/jasa/info/authors and then clicking on the item *List of Phonetic Symbols*. This file gives, for each symbol (displayed in 3 different Unicode fonts, DoulosSIL, GentiumPlus, and CharisSILCompact): its Unicode hex ID number, the Unicode character set it is part of, its Unicode character name, and its IPA definition (taken from the IPA chart). Most of these symbols and their Unicode numbers are also available from Professor John Wells of University College London at http://www.phon.ucl.ac.uk/home/wells/ipa-unicode.htm#alfa, without the Unicode character names and character set names.

The method of including such symbols in a manuscript is to use, in conjunction with a word processor, a Unicode-compliant font that includes all symbols required. Fonts that are not Unicode-compliant should not be used. Most computers come with Unicode fonts that give partial coverage of the IPA. Some sources where one can obtain Unicode fonts for Windows, MacOS, and Linux with full IPA coverage are http://www.phon.ucl.ac.uk/home/wells/ipa-unicode.htm and http://scripts.sil.org/cms/scripts/page.php?item_id=SILFontList. Further information about which fonts contain a desired symbol set can be found at http://www.alanwood.net/unicode/fontsbyrange.html#u0250 and adjacent pages at that site. While authors may use any Unicode-compliant font in their manuscript, AIP Publishing reserves the right to replace the author's font with a Unicode font of its choice (currently one of the SIL fonts Doulos, Gentium, or Charis, but subject to change in the future).

For LaTeX manuscripts, EM's LaTeX-processing environment (MikTeX) supports the use of TIPA fonts. TIPA fonts are available through the Comprehensive TeX Archive Network at http://www.ctan.org/ (download from http://www.ctan.org/pkg/tipa).

H. Figures

Each figure should be manifestly legible when reduced to one column of the printed journal page. Figures requiring the full width of a journal page are discouraged, but exceptions can be made if the reasons for such are sufficiently evident. The inclusion of figures in the manuscript should be such that the manuscript, when published, should ordinarily have no more than 30% of the space devoted to figures, and such that the total number of figures should ordinarily not be more than 20. In terms of the restriction of the total space for figures, each figure part will be considered as occupying a quarter page.

The figures are numbered in the order in which they are first referred to in the text. There must be one such referral for every figure in the text. Each figure must have a caption, and the captions are gathered together into a single list that appears at the end of the manuscript.

A chief criticism of many contemporary papers is that they contain far too many computer-generated graphical illustrations that present numerical results. An author develops a certain general computational method (realized by software) and then uses it to exhaustively discuss a large number of special cases. This practice must be avoided. Unless there is an overwhelmingly important single point that the sequence of figures demonstrates as a whole, an applicable rule of thumb is that the maximum number of figures of a given type must be four.

The clarity of most papers is greatly improved if the authors include one or more explanatory sketches. If, for example, the mathematical development presumes a certain geometrical arrangement, then a sketch of this arrangement must be included in the manuscript. If the experiment is carried out with a certain setup of instrumentation and apparatuses, then a sketch is also appropriate.

Color figures can be included in the online version of the *Journal* with no extra charge provided that these appear suitably as black and white figures in the print edition.

I. Tables

Tables are numbered by capital roman numerals (TABLE III, TABLE IV, etc.) and are collected at the end of the manuscript, following the references and preceding the figure captions, one table per page. There should be a descriptive caption (not a title) above each table in the manuscript.

Footnotes to individual items in a table are designated by raised lowercase letters (0.123[a], Martin[b], etc.) The footnotes as such are given below the table and should be as brief as practicable. If the footnotes are to references already cited in the text, then they should have forms such as— [a]Reference 10—or—[b]Firestone (1935)—depending on the citation style adopted in the text. If the reference is not cited in the text, then the footnote has the same form as a textual footnote when the alphabetical bibliographic list style is used. One would cast the footnote as in the second example above and then include a reference to a 1935 work by Firestone in the paper's overall bibliographic list. In general, it is recommended that no footnote refer to references that are not already cited in the text.

VI. THE COVER LETTER

The contents of the cover letter are usually perfunctory. There are, however, some circumstances where material in a cover letter file might be advisable or needed:

(1) If persons who would ordinarily have been included as authors have given permission or requested that their names not be included, then that must be so stated. (This requirement is imposed because some awkward situations have arisen in the past in which persons have complained that colleagues or former colleagues have deliberately omitted their names as authors from papers to which they have contributed. The *Journal* also has the policy that a paper may still be published, even if one of the persons who has contributed to the work refuses to allow his or her name to be included among the list of authors, providing there is no question of plagiarism.)

Unless a cover letter listing such exceptions is submitted, the submittal process implies that the corresponding author is attesting that the author list is complete.

(2) If there has been any prior presentation or any overlap in concept with any other manuscripts that have been either published or submitted for publication, this must be stated in a cover letter. If the manuscript has been previously submitted elsewhere for publication, and subsequently withdrawn, this must also be disclosed. If none of these apply for the submitted manuscript, then the submission process is construed to imply that the corresponding author is attesting to such a fact.

(3) (Optional.) Reasons why the authors have selected to submit their paper to JASA rather than some other journal. These would ordinarily be supplied if the authors are concerned that there may be some questions as to the paper meeting the "truly acoustics" criterion or of its being within the scope of the *Journal.* If none of the references cited in the submitted paper are to articles previously published in the *Journal,* it is highly advisable that some strong reasons be given for why the authors believe the paper falls within the scope of the *Journal.*

(4) If the online submission includes the listing of one or more persons who the authors prefer not be used as reviewers, an explanation in a cover letter would be desirable.

(5) If the authors wish to make statements which they feel are appropriate to be read by editors, but are inappropriate to be included in the actual manuscript, then such should be included in a cover letter.

Cover letters are treated by the EM system as being distinct from *rebuttal letters.*

Rebuttal letters should be submitted with revised manuscripts, and the contents are usually such that the authors give, when appropriate, rebuttals to suggestions and criticisms of the reviewers, and give detailed discussion of how and why the revised manuscript differs from what was originally submitted.

VII. EXPLANATIONS AND CATEGORIES

A. Suggestions for Associate Editors

In the suggestion of an Associate Editor who should handle a specific manuscript, authors should consult a document titled "Associate Editors identified with PACS classification items" obtainable at the JASA web site <http://scitation.aip.org/content/asa/journal/jasa/info/about>. Here the Associate Editors are identified by their initials, and the relation of the initials to the names is easily discerned from the listing of Associate Editors on the back cover of each issue, on the title page of each volume, and at the online site <http://scitation.aip.org/content/asa/journal/jasa/info/about>.

Authors are not constrained to select Associate Editors specifically identified with their choice of principal ASA Technical Committee and should note that the *Journal* has special Associate Editors for Mathematical Acoustics, Computational Acoustics, and Education in Acoustics. Review, forum, and tutorial articles are ordinarily invited; submission of unsolicited review articles, forum articles, or tutorial articles (other than those which can be construed as papers on education in acoustics) without prior discussion with the Editor-in-Chief is discouraged. Authors should suggest the Associate Editor for Education in Acoustics for tutorial papers that contain material which might be used in standard courses on acoustics or material that supplements standard textbooks.

B. Types of manuscripts

Categories of papers that are published in the *Journal* include the following:

1. Regular research articles

These are papers which report original research. There is neither a lower limit nor an upper limit on their length, although authors must pay page charges if the length results in more than 12 printed pages. The prime requirement is that such papers must contain a complete account of the reported research. These articles are classified in JASA's Table of Contents by their (most appropriate) Technical Committee or by Education in Acoustics.

2. Letters to the Editor

These are shorter research contributions that can be any of the following: (i) an announcement of a research result, preliminary to the full of the research; (ii) a scientific or technical discussion of a topic that is timely; (iii) brief alternate derivations or alternate experimental evidence concerning acoustical phenomena; (iv) provocative articles that may stimulate further research. Brevity is an essential feature of a letter, and the *Journal* suggests 3 printed journal pages as an upper limit, although it will allow up to 4 printed pages in exceptional cases.

The *Journal's* current format has been chosen so as to give letters greater prominence. Their brevity in conjunction with the possible timeliness of their contents gives impetus to a quicker processing and to a shorter time lag between submission and appearance in printed form in the *Journal.* (The quickest route to publication that the Acoustical Society currently offers is submission to the special section *JASA Express Letters* (JASA-EL) of the *Journal.* For information regarding JASA-EL, visit the site <http://scitation.aip.org/content/asa/journal/jasael/info/authors>.)

Because the desire for brevity is regarded as important, the author is not compelled to make a detailed attempt to place the work within the context of current research; the citations are relatively few and the review of related research is limited. The author should have some reason for desiring a more rapid publication than for a normal article, and the editors and the reviewers should concur with this. The work should have a modicum of completeness, to the extent that the letter "tells a story" that is at least plausible to the reader, and it should have some nontrivial support for what is being related. Not all the loose strings need be tied together. Often there is an implicit promise that the publication of the letter will be followed up by a regular research article that fills in the gaps and that does all the things that a regular research article should do.

3. Errata

These must be corrections to what actually was printed. Authors must explicitly identify the passages or equations in the paper and then state what should replace them. Long essays on why a mistake was made are not desired. A typical line in an errata article would be of the form: *Equation (23) on page 6341 is incorrect. The correct version is ...* . For detailed examples, the authors should look at previously published errata articles in the *Journal.*

4. Comments on published papers

Occasionally, one or more readers, after reading a published paper, will decide to submit a paper giving comments about that paper. The *Journal* welcomes submissions of this type, although they are reviewed to make sure that the comments are reasonable and that they are free of personal slurs. The format of the title of a comments paper is rigidly prescribed, and examples can be found in previous issues of the *Journal.* The authors of the papers under criticism are frequently consulted as reviewers, but their unsubstantiated opinion as to whether the letter is publishable is usually not given much weight.

5. Forum letters

Forum letters are analogous to the "letters to the editor" that one finds in the editorial section of major newspapers. They may express opinions or advocate actions. They may also relate anecdotes or historical facts that may be of general interest to the readers of the *Journal.* They need not have a title and should not have an abstract; they also should be brief, and they should not be of a highly technical nature. These are also submitted using the EM system, but are not handled as research articles. The applicable Associate Editor is presently the Editor-in-Chief. For examples of acceptable letters and the format that is desired, prospective authors of such letters should consult examples that have appeared in recent issues of the *Journal.*

6. Tutorial and review papers

Review and tutorial papers are occasionally accepted for publication, but are difficult to handle within the peer-review process. All are handled directly by the Editor-in-Chief, but usually with extensive discussion with the relevant Associate Editors. Usually such are invited, based on recommendations from the Associate Editors and the Technical Committees of the Society, and the tentative acceptance is based on a submitted outline and on the editors' acquaintance with the prospective author's past work. The format of such papers is similar to those of regular research articles, although there should be a table of contents following the abstract for longer research articles. Submission is handled by the online system, but the cover letter should discuss the history of prior discussions with the editors. Because of the large expenditure of time required to write an authorative review article, authors are advised not to begin writing until they have some assurance that there is a good likelihood of the submission eventually being accepted.

7. Book reviews

All book reviews must be first invited by the Associate Editor responsible for book reviews. The format for such reviews is prescribed by the Associate Editor, and the EM submittal process is used primarily to facilitate the incorporation of the reviews into the Journal.

8. Special Issues

A Special Issue must be proposed to the Editor-in-Chief by a person who is willing and able to work as a Guest Editor or coordinator, along with a regular Associate Editor. Such issues are encouraged (though not strictly required) to have an open call for papers, which will be posted on ASA's Scitation web page. Time limits for submission, review, and revision are usually enforced. If the total Special Issue is less than 100 Journal pages long, it will be printed as a part of the current Journal volume, rather than separately. Special Issues are a definite attraction for readers, and good ideas for Special Issues are always welcome.

9. Guest Invited Articles

In order to solicit papers of interest to acousticians, but outside the normal range of topics found in JASA, we have initiated the category of Guest Invited Article. These must be approved by the Editor-in-Chief, but suggestions are welcome from all ASA members.

10. Addenda

In rare cases, a small addendum may be submitted to augment a paper on a key or critical point. These are not encouraged, but can be submitted if a good case for their need is made to the Editor-in-Chief.

11. Retractions

Again, in the rare case an article has a fatal flaw, an author can contact the Editor-in-Chief about a possible retraction of the article's content. (The original article will still be part of the permanent record.)

12. Other submission categories

There are several article categories that appear on the FM submission list that are reserved for journal personnel use, and are not for general submissions. They are: Calendar, Thank You to Reviewers, Technical Committee Reports, Acoustical News, Acoustical Standards News, Reviews of Acoustical Patents, and Editorial.

VIII. FACTORS RELEVANT TO PUBLICATION DECISIONS

A. Peer review system

The *Journal* uses a peer review system in the determination of which submitted manuscripts should be published. The Associate Editors make the actual decisions; each editor has specialized understanding and prior distinguished accomplishments in the subfield of acoustics that encompasses the contributed manuscript. They seek advice from reviewers who are knowledgeable in the general subject of the paper, and the reviewers give opinions on various aspects of the work; primary questions are whether the work is original and whether it is correct. The Associate Editor and the reviewers who examine the manuscript are the authors' peers: persons with comparable standing in the same research field as the authors themselves. (Individuals interested in reviewing for JASA or for JASA-EL can convey that interest via an e-mail message to the Editor-in-Chief at <jasa@aip.org>.)

B. Selection criteria

Many submitted manuscripts are not selected for publication. Selection is based on the following factors: adherence to the stylistic requirements of the *Journal,* clarity of exposition, originality of the contribution, demonstrated understanding of previously published literature pertaining to the subject matter, appropriate discussion of the relationships of the reported research to other current research or applications, appropriateness of the subject matter to the *Journal,* correctness of the content of the article, completeness of the reporting of results, the reproducibility of the results, and the significance of the contribution. The *Journal* reserves the right to refuse publication of any submitted article without giving extensively documented reasons, although the editors usually give suggestions that can help the authors in the writing and submission of future papers. The Associate Editor also has the option, but not an obligation, of giving authors an opportunity to submit a revised manuscript addressing specific criticisms raised in the peer review process. The selection process occasionally results in mistakes, but the time limitations of the editors and the reviewers preclude extraordinary steps being taken to ensure that no mistakes are ever made. If an author feels that the decision may have been affected by an *a priori* adverse bias (such as a conflict of interest on the part of one of the reviewers), the system allows authors to express the reasons in writing and ask for an appeal review.

C. Scope of the *Journal*

Before one decides to submit a paper to the *Journal of the Acoustical Society*, it is prudent to give some thought as to whether the paper falls within the scope of the *Journal*. While this can in principle be construed very broadly, it is often the case that another journal would be a more appropriate choice. As a practical matter, the *Journal* would find it difficult to give an adequate peer review to a submitted manuscript that does not fall within the broader areas of expertise of any of its Associate Editors. In the *Journal's* peer-review process, extensive efforts are made to match a submitted manuscript with an Associate Editor knowledgeable in the field, and the Editors have the option of declining to take on the task. It is a tacit understanding that no Associate Editor should accept a paper unless he or she understands the gist of the paper and is able to make a knowledgeable assessment of the relevance of the advice of the selected reviewers. If no one wishes to handle a manuscript, the matter is referred to the Editor-in-Chief and a possible resulting decision is that the manuscript is outside the de facto scope of the *Journal.*

The *Journal* ordinarily selects for publication only articles that have a clear identification with acoustics. It would, for example, not ordinarily publish articles that report results and techniques that are not specifically applicable to acoustics, even though they could be of interest to some persons whose work is concerned with acoustics. An editorial[3] published in the October 1999 issue gives examples that are *not* clearly identifiable with acoustics.

IX. POLICIES REGARDING PRIOR PUBLICATION

The *Journal* adheres assiduously to all applicable copyright laws, and authors must not submit articles whose publication will result in a violation of such laws. Furthermore, the *Journal* follows the tradition of providing an orderly archive of scientific research in which authors take care that results and ideas are fully attributed to their originators. Conscious plagiarism is a serious breach of ethics, if not illegal. (Submission of an article that is plagiarized, in part or in full, may have serious repercussions on the future careers of the authors.) Occasionally, authors rediscover older results and submit papers reporting these results as though they were new. The desire to safeguard the *Journal* from publishing any such paper requires that submitted articles have a sufficient discussion of prior related literature to demonstrate the authors' familiarity with the literature and to establish the credibility of the assertion that the authors have carried out a thorough literature search.

In many cases, the authors themselves may have either previously circulated, published, or presented work that has substantial similarities with what is contained within the contributed manuscript. In general, JASA will not publish work that has been previously published. (An exception is when the previous publication is a letter to the editor, and when pertinent details were omitted because of the brief nature of the earlier reporting.) Presentations at conferences are not construed as prior publication; neither is the circulation of preprints or the posting of preprints on any web site, providing the site does not have the semblance of an archival online journal. Publication as such implies that the work is currently, and for the indefinite future, available, either for purchase or on loan, to a broad segment of the research community. Often the *Journal* will consider publishing manuscripts with tangible similarities to other work previously published by the authors—providing the following conditions are met: (1) the titles are different; (2) the submitted manuscript contains no extensive passages of text or figures that are the same as in the previous publication; (3) the present manuscript is a substantial update of the previous publication; (4) the previous publication has substantially less availability than would a publication in JASA; (5) the current manuscript gives ample referencing to the prior publication and explains how the current manuscript differs from the prior publication. Decisions regarding such cases are made by the Associate Editors, often in consultation with the Editor-in-Chief. (Inquiries prior to submission as to whether a given manuscript with some prior history of publication may be regarded as suitable for JASA should be addressed to the Editor-in-Chief at <jasa@aip.org>.)

The *Journal* will not consider any manuscript for publication that is presently under consideration by another journal or which is substantially similar to another one under consideration. If it should learn that such is the case, the paper will be rejected and the editors of the other journal will be notified.

Authors of an article previously published as a letter to the editor, either as a regular letter or as a letter in the JASA-EL (*JASA Express Letters*) section of the *Journal,* where the original account was either abbreviated or preliminary are encouraged to submit a more comprehensive and up-dated account of their research to the *Journal.* The same holds for POMA (Proceedings of Meetings on Acoustics) articles.

A. Speculative papers

In some cases, a paper may be largely speculative; a new theory may be offered for an as yet imperfectly understood phenomenon, without complete confirmation by experiment. Although such papers may be controversial, they often become the most important papers in the long-term development of a scientific field. They also play an important role in the stimulation of good research. Such papers are intrinsically publishable in JASA, although explicit guidelines for their selection are difficult to formulate. Of major importance are (i) that the logical development be as complete as practicable, (ii) that the principal ideas be plausible and consistent with what is currently known, (iii) that there be no known counter-examples, and (iv) that the authors give some hints as to how the ideas might be checked by future experiments or numerical computations. In addition, the authors should cite whatever prior literature exists that might indicate that others have made similar speculations.

B. Multiple submissions

The current online submittal process requires that each paper be submitted independently. Each received manuscript will be separately reviewed and judged regarding its merits for publication independently of the others. There is no formal mechanism for an author to request that two submissions, closely spaced in their times of submission, be regarded as a single submission.

In particular, the submission of two manuscripts, one labeled "Part I" and the other labeled "Part II" is not allowed. Submission of a single manuscript with the label "Part I" is also not allowed. An author may submit a separate manuscript labeled "Part II," if the text identifies which previously accepted paper is to be regarded as "Part I." Doing so may be a convenient method for alerting potential readers to the fact that the paper is a sequel to a previous paper by the author. The author should not submit a paper so labeled, however, unless the paper to be designated as "Part I" has already been accepted, either for JASA or another journal.

The Associate Editors are instructed not to process any manuscript that cannot be read without the help of as yet unpublished papers that are still under review. Consequently, authors are requested to hold back the submission of "sequels" to previously submitted papers until the disposition of those papers is determined. Alternately, authors should write the "sequels" so that the reading and comprehension of those manuscripts does not require prior reading and access of papers whose publication is still uncertain.

X. SUGGESTIONS REGARDING CONTENT

A. Introductory section

Every paper begins with introductory paragraphs. Except for short Letters to the Editor, these paragraphs appear within a separate principal section, usually with the heading "Introduction."

Although some discussion of the background of the work may be advisable, a statement of the precise subject of the work must appear within the first two paragraphs. The reader need not fully understand the subject the first time it is stated; subsequent sentences and paragraphs should clarify the statement and should supply further necessary background. The extent of the clarification must be such that a nonspecialist will be able to obtain a reasonable idea of what the paper is about. The Introduction should also explain to the nonspecialist just how the present work fits into the context of other current work done by persons other than the authors themselves. Beyond meeting these obligations, the writing should be as concise as practicable.

The Introduction must give the authors' best arguments as to why the work is original and significant. This is customarily done via a knowledgeable discussion of current and prior literature. The authors should envision typical readers or typical reviewers, and this should be a set of people that is not inordinately small, and the authors must write so as to convince them. In some cases, both originality and significance will be immediately evident to all such persons, and the arguments can be brief. In other cases, the authors may have a daunting task. It must not be assumed that readers and reviewers will give the authors the benefit of the doubt.

B. Main body of text

The writing in the main body of the paper must follow a consistent logical order. It should contain only material that pertains to the main premise of the paper, and that premise should have been stated in the Introduction. While tutorial discussions may in some places be appropriate, such should be kept to a minimum and should be only to the extent necessary to keep the envisioned readers from becoming lost.

The writing throughout the text, including the Introduction, must be in the present tense. It may be tempting to refer to subsequent sections and passages in the manuscript in the future tense, but the authors must assiduously avoid doing so, using instead phrases such as "is discussed further below."

Whenever pertinent results, primary or secondary, are reached in the progress of the paper, the writing should point out that these are pertinent results in such a manner that it would get the attention of a reader who is rapidly scanning the paper.

The requirement of a consistent logical order implies that the logical steps appear in consecutive order. Readers must not be referred to subsequent passages or to appendixes to fill in key elements of the logical development. The fact that any one such key element is lengthy or awkward is insufficient reason to relegate it to an appendix. Authors can, however, flag such passages giving the casual reader the option of skipping over them on first reading. The writing nevertheless must be directed toward the critical reader—a person who accepts no aspect of the paper on faith. (If the paper has some elements that are primarily speculative, then that should be explicitly stated, and the development should be directed toward establishing the plausibility of the speculation for the critical reader.)

To achieve clarity and readability, the authors must explicitly state the purposes of lengthy descriptions or of lengthy derivations at the beginning of the relevant passages. There should be no mysteries throughout the manuscript as to the direction in which the presentation is going.

Authors must take care that no reader becomes needlessly lost because of the use of lesser-known terminology. All terms not in standard dictionaries must be defined when they are first used. Acronyms should be avoided, but, when they are necessary, they must be explicitly defined when first used. The terminology must be consistent; different words should not be used to represent the same concept.

Efforts must be taken to avoid insulting the reader with the use of gratuitous terms or phrases such as *obvious, well-known, evident,* or *trivial.* If the adjectives are applicable, then they are unnecessary. If not, then the authors risk incurring the ill-will of the readers.

If it becomes necessary to bring in externally obtained results, then the reader must be apprised, preferably by an explicit citation to accessible literature, of the source of such results. There must be no vague allusions, such as "It has been found that..." or "It can be shown that..." If the allusion is to a mathematical derivation that the authors have themselves carried out, but which they feel is not worth describing in detail, then they should briefly outline how the derivation can be carried out, with the implication that a competent reader can fill in the necessary steps without difficulty.

For an archival journal such as JASA, reproducibility of reported results is of prime importance. Consequently, authors must give a sufficiently detailed account, so that all results, other than anecdotal, can be checked by a competent reader with comparable research facilities. If the results are numerical, then the authors must give estimates of the probable errors and state how they arrived at such estimates. (Anecdotal results are typically results of field experiments or unique case studies; such are often worth publishing as they can stimulate further work and can be used in conjunction with other results to piece together a coherent understanding of broader classes of phenomena.)

C. Concluding section

The last principal section of the article is customarily labeled "Conclusions" or "Concluding Remarks." This should not repeat the abstract, and it should not restate the subject of the paper. The wording should be directed toward a person who has some, if not thorough, familiarity with the main body of the text and who knows what the paper is all about. The authors should review the principal results of the paper and should point out just where these emerged in the body of the text. There should be a frank discussion of the limitations, if any, of the results, and there should be a broad discussion of possible implications of these results.

Often the concluding section gracefully ends with speculations on what research might be done in the future to build upon the results of the present paper. Here the authors must write in a collegial tone. There should be no remarks stating what the authors themselves intend to do next. They must be careful not to imply that the future work in the subject matter of the paper is the exclusive domain of the authors, and there should be no allusions to work in progress or to work whose publication is uncertain. It is conceivable that readers stimulated to do work along the lines suggested by the paper will contact the authors directly to avoid a duplication of effort, but that will be their choice. The spirit expressed in the paper itself should be that anyone should be free to follow-up on the suggestions made in the concluding section. A successful paper is one that does incite such interest on the part of the readers and one which is extensively cited in future papers written by persons other than the authors themselves.

D. Appendixes

The *Journal* prefers that articles not include appendixes unless there are strong reasons for their being included. Details of mathematical developments or of experimental procedures that are critical to the understanding of the substance of a paper must not be relegated to an appendix. (Authors must bear in mind that readers can easily skim over difficult passages in their first reading of a paper.) Lengthy proofs of theorems may possibly be placed in appendixes providing their stating as such in the main body of the text is manifestly plausible. Short appendixes are generally unnecessary and impede the comprehension of the paper. Appendixes may be used for lengthy tabulations of data, of explicit formulas for special cases, and of numerical results. Editors and reviewers, however, may question whether their inclusion is necessary.

E. Selection of references

References are typically cited extensively in the Introduction, and the selection of such references can play an important role in the potential usefulness of the paper to future readers and in the opinions that readers and reviewers form of the paper. No hard and fast rules can be set down as to how authors can best select references and as to how they should discuss them, but some suggestions can be found in an editorial[4] published in the May 2000 issue. If a paper falls within the scope of the *Journal,* one would ordinarily expect to find several references to papers previously published in JASA.

Demonstration of the relevance of the work is often accomplished via citations, with accompanying discussion, to recent articles in JASA and analogous journals. The implied claims to originality can be strengthened via citations, with accompanying discussion, to prior work related to the subject of the paper, sufficient to establish credibility that the authors are familiar with the literature and are not duplicating previous published work. Unsupported assertions that the authors are familiar with all applicable literature and that they have carried out an exhaustive literature survey are generally unconvincing to the critical reader.

Authors must not make large block citations of many references (e.g., four or more). There must be a stated reason for the citation of each reference, although the same reason can sometimes apply simultaneously to a small number of references. The total number of references should be kept as small a number as is consistent with the principal purposes of the paper (45 references is a suggested upper limit for a regular research article). Although nonspecialist readers may find a given paper to be informative in regard to the general state of a given field, the authors must not consciously write a research paper so that it will fulfill a dual function of being a review paper or of being a tutorial paper.

Less literate readers often form and propagate erroneous opinions concerning priority of ideas and discoveries based on the reading of recent papers, so authors must make a conscious attempt to cite original sources. Secondary sources can also be cited, if they are identified as such and especially if they are more accessible or if they provide more readable accounts. In such cases, reasons must be given as to why the secondary sources are being cited. References to individual textbooks for results that can be found in a large number of analogous textbooks should not be given, unless the cited textbook gives a uniquely clear or detailed discussion of the result. Authors should assume that any reader has access to some such textbook, and the authors should tacitly treat the result as well-known and not requiring a reference citation.

Authors must not cite any reference that the authors have not explicitly seen, unless the paper has a statement to that effect, accompanied by a statement of how the authors became aware of the reference. Such citations should be limited to crediting priority, and there must be no implied recommendations that readers should read literature which the authors themselves have not read.

F. Multimedia

A benefit of publishing in an electronic online journal is the ability to integrate multimedia files into both the published and archived articles. The online presentation of the paper allows for links to both audio and video clips directly from within the text of the article. The multimedia files submitted for *JASA* will be reviewed as part of the peer review process and accepted for publication in much the same way as are two-dimensional figures for traditional print journals. The multimedia submission guidelines presented here are subject to change because of improvements and increasing availability of the relevant technology.

The implementation of *JASA* on the Editorial Manager system is such that multimedia files are submitted in the same manner as are figure files, i.e., they are uploaded individually during the manuscript submission process. The sequence in which they are uploaded should be the same as that in which they are referred to in the text. The text should refer to these files using the designations Mm. 1, Mm. 2, etc.; this is similar to the convention of referring to figures as Fig. 1, Fig. 2, etc.

To ensure broad viewing/playing ability across hardware platforms and browsers, submissions in a variety of file formats are acceptable.

Acceptable video formats are: (i) QuickTime movies (mov); (ii) Mpeg movies (mpg); (iii) Animated Gifs (gif); (iv) Audio Video Interleave (avi).

Acceptable audio formats are: (i) AIFF (aif); (ii) Wav (wav); (iii) MP3 (mp3) at 128 kB or greater.

In the above lists, the letters in parentheses are the standard suffixes for files in the corresponding format. For example, fancymovie.mov would be a file containing a QuickTime movie.

It is important that authors make their multimedia files no larger or numerous than necessary to convey scientific information that is central to the manuscript's purpose. Authors should consider that files larger than several MB are problematic for readers using dial-up connections. Files larger than 10 MB require permission from the Editor. When video compression is used, the codec software module must be widely available. Files may not be compressed into archives, such as .zip and .tar formats. Since readers may find it tedious to download numerous files that contribute little new information, authors must select their materials carefully. Submissions with more than 6 multimedia files must receive permission from the Editor.

In the typesetting of an accepted manuscript, links will be placed within the online publication for each of the multimedia files. During the peer-review process, the reviewers and editors will access such files by going to the online site reserved for the submitted manuscript and its accompanying files, and then selecting whatever multimedia file is desired.

To help the publisher in determining just where links to each multimedia file are to be placed, authors should give a multimedia caption following the first paragraph in which the file is mentioned. The multimedia caption should resemble the following example:

Mm. 2. Fancy video file. This is a file of type "mov" (1.2 Mb).

Here "Fancy video file" is the caption for the multimedia object, which contains a level of description similar to a figure or table caption. The primary purpose of including the file type and its size is to allow readers to determine whether they wish to download it.

Authors may also wish to have a figure included for each of the video files that accompany the manuscript. One way of doing this is to take a single frame and convert it to a figure file, and then treat this in the same way as one would treat any other figure. However, the caption for such a figure should refer to the Mm number of the corresponding video file and should give a brief description of what can be found in that file.

XI. SUGGESTIONS REGARDING STYLE

A. Quality of writing and word usage

The *Journal* publishes articles in the English language only. There are very few differences of substance between British English style (as codified in the *Oxford English Dictionary*[5]) and US English style, but authors frequently must make choices in this respect, such as between alternate spelling of words that end in either *-or* or *-our,* or in either *-ized* or *-ised,* or in either *-er* or *-re.*

Articles published in JASA are expected to adhere to high standards of scholarly writing. A formal writing style free of slang is required. Good conversational skills do not necessarily translate to good formal writing skills. Authors are expected to make whatever use is necessary of standard authoritative references in regard to English grammar and writing style in preparing their manuscripts. Many good references exist—among those frequently used by professional writers are Webster's Third New International Dictionary, Unabridged,[6] Merriam-Webster's Collegiate Dictionary, 11th Edition,[7] Strunk and White's Elements of Style,[8] and the Chicago Manual of Style.[9] (The Third New International is AIP Publishing's standard dictionary.) All authors are urged to do their best to produce a high quality readable manuscript, consistent with the best traditions of scholarly and erudite writing. Occasional typographical errors and lapses of grammar can be taken care of in the copy-editing phase of the production process. Receipt of a paper whose grammatical and style errors are so excessive that they cannot be easily fixed by copy-editing will generally result in the authors being notified that the submission is not acceptable. Receipt of such a notification should not be construed as a rejection of the manuscript—the authors should take steps, possibly with external help, to revise the manuscript so that it overcomes these deficiencies. (Authors needing help or advice on scientific writing in the English language are encouraged to contact colleagues, both within and outside their own institutions, to crititque the writing in their manuscripts. Unfortunately, the staff of the *Journal* does not have the time to do this on a routine basis.)

There are some minor discrepancies in the stylistic rules that are prescribed in various references—these generally arise because of the differences in priorities that are set in different publication categories. Newspapers, for example, put high emphasis on the efficient use of limited space for conveying the news and for catching the interest of their readers. For scholarly journals, on the other hand, the overwhelming priority is clarity. In the references cited above, this is the basis for most of the stated rules. In following this tradition, the *Journal,* for example, requires a rigorous adherence to the serial comma rule (Strunk's rule number 2): In a series of three or more terms with a single conjunction, use a comma after each term except the last. Thus a JASA manuscript would refer to the "theory of Rayleigh, Helmholtz, and Kirchhoff" rather than to the "theory of Rayleigh, Helmholtz and Kirchhoff."

The priority of clarity requires that authors only use words that are likely to be understood by a large majority of potential readers. Usable words are those whose definitions may be found either in a standard unabridged English dictionary (such as the Webster's Third New International mentioned above), in a standard scientific dictionary such as the Academic Press Dictionary of Science and Technology,[10] or in a dictionary specifically devoted to acoustics such as the Dictionary of Acoustics[11] by C. L. Morfey. In some cases, words and phrases that are not in any dictionary may be *in vogue* among some workers in a given field, especially among the authors and their colleagues. Authors must give careful consideration to whether use of such terms in their manuscript is necessary; and if the authors decide to use them, precise definitions must be stated within the manuscript. Unilateral coinage of new terms by the authors is discouraged. In some cases, words with different meanings and with different spellings are pronounced exactly the same, and authors must be careful to choose the right spelling. Common errors are to interchange "principal" and "principle" and to interchange "role" and "roll."

B. Grammatical pitfalls

There are only a relatively small number of categories of errors that authors frequently make in the preparation of manuscripts. Authors should be aware of these common pitfalls and double-check that their manuscripts contain no errors in these categories. Some errors will be evident when the manuscript is read aloud; others, depending on the background of the writers, may not be. Common categories are (1) dangling participles, (2) lack of agreement in number (plural versus singular) of verbs with their subjects, (3) omission of necessary articles (such as "a," "an," and "the") that precede nouns, (4) the use of incorrect case forms (subjective, objective, possessive) for pronouns (e.g., who versus whom), and (5) use of the incorrect form (present, past, past participle, and future) in regard to tense for a verb. Individual authors may have their own peculiar pitfalls, and an independent casual reading of the manuscript by another person will generally pinpoint such pitfalls. Given the recognition that such exist, a diligent author should be able to go through the manuscript and find all instances where errors of the identified types occur.

C. Active voice and personal pronouns

Many authorities on good writing emphasize that authors should use the active rather than the passive voice. Doing so in scholarly writing, especially when mathematical expressions are present, is often infeasible, but the advice has merit. In mathematical derivations, for example, some authors use the tutorial "we" to avoid using the passive voice, so that one writes: "We substitute the expression on the right side of Eq. (5) into Eq. (2) and obtain ...," rather than: "The right side of Eq. (5) is substituted into Eq. (2), with the result being ... ." A preferable construction is to avoid the use of the tutorial "we" and to use transitive verbs such as "yields," "generates," "produces," and "leads to." Thus one would write the example above as: "Substitution of Eq. (5) into Eq. (2) yields ... ." Good writers frequently go over an early draft of a manuscript, examine each sentence and phrase written using the passive voice, and consider whether they can improve the sentence by rewriting it.

In general, personal pronouns, including the "tutorial we," are preferably avoided in scholarly writing, so that the tone is impersonal and dispassionate. In a few cases, it is appropriate that an opinion be given or that a unique personal experience be related, and personal pronouns are unavoidable. What should be assiduously avoided are any egotistical statements using personal pronouns. If a personal opinion needs to be expressed, a preferred construction is to refer to the author in the third person, such as: "the present writer believes that ... ."

D. Acronyms

Acronyms have the inconvenient feature that, should the reader be unfamiliar with them, the reader is clueless as to their meaning. Articles in scholarly journals should ideally be intelligible to many generations of future readers, and formerly common acronyms such as RCA (Radio Corporation of America, merged into the General Electric Corporation) and REA (Rural Electrification Authority) may have no meaning to such readers. Consequently, authors are requested to use acronyms sparingly and generally only when not using them would result in exceedingly awkward prose. Acronyms, such as SONAR and LASER (currently written in lowercase, sonar and laser, as ordinary words), that have become standard terms in the English language and that can be readily found in abridged dictionaries, are exceptions. If the authors use acronyms not in this category, then the meaning of the individual letters should be spelled out at the time such an acronym is first introduced.

E. Computer programs

In some cases the archival reporting of research suggests that authors give the names of specific computer programs used in the research. If the computation or data processing could just as well have been carried out with the aid of any one of a variety of such programs, then the name should be omitted. If the program has unique features that are used in the current research, then the stating of the program name must be accompanied by a brief explanation of the principal premises and functions on which the relevant features are based. One overriding consideration is that the *Journal* wishes to avoid implied endorsements of any commercial product.

F. Code words

Large research projects and large experiments that involve several research groups are frequently referred to by code words. Research articles in the *Journal* must be intelligible to a much broader group of readers, both present and future, than those individuals involved in the projects with which such a code word is associated. If possible, such code words should either not be used or else referred to in only a parenthetical sense. If attempting to do this leads to exceptionally awkward writing, then the authors must take special care to explicitly explain the nature of the project early in the paper. They must avoid any impression that the paper is specifically directed toward members of some in-group.

REFERENCES

[1]M. Mellody and G. H. Wakefield, "The time-frequency characteristics of violin vibrato: Modal distribution analysis and synthesis," J. Acoust. Soc. Am. **107,** 598–611 (2000).

[2]See, for example, the paper: B. Møhl, M. Wahlberg, P. Madsen, L. A. Mller, and A. Surlykke, "Sperm whale clicks: Directionality and source level revisited," J. Acoust. Soc. Am. **107,** 638–648 (2000).

[3]A. D. Pierce, "Current criteria for selection of articles for publication," J. Acoust. Soc. Am. **106,** 1613–1616 (1999).

[4]A. D. Pierce, "Literate writing and collegial citing," J. Acoust. Soc. Am. **107,** 2303–2311 (2000).

[5]*The Oxford English Dictionary,* 2nd ed., edited by J. Simpson and E. Weiner (Oxford University Press, 1989), 20 volumes. Also published as *Oxford English Dictionary (Second Edition) on CD-ROM, version 2.0* (Oxford University Press, 1999). An online version is available by subscription at the Internet site http://public.oed.com/about/free-oed/.

[6]*Webster's Third New International Dictionary of the English Language, Unabridged,* Philip Babcock Gove, Editor-in-Chief (Merriam-Webster Inc., Springfield, MA, 1993, principal copyright 1961). This is the eighth in a series of dictionaries that has its beginning in Noah Webster's *American Dictionary of the English Language* (1828).

[7]*Merriam-Webster's Collegiate Dictionary, 11th Edition* (Merriam-Webster, Springfield, MA, 2003, principal copyright 1993). (A freshly updated version is issued annually.)

[8]W. Strunk, Jr. and E. B. White, *The Elements of Style,* 4th ed., with forward by Roger Angell (Allyn and Bacon, 1999).

[9]*The Chicago Manual of Style: The Essential Guide for Writers, Editors, and Publishers,* 14th ed., with preface by John Grossman (University of Chicago Press, 1993).

[10]*Academic Press Dictionary of Science and Technology,* edited by Christopher Morris (Academic Press, Inc., San Diego, 1992).

[11]C. L. Morfey, *Dictionary of Acoustics* (Academic Press, Inc., San Diego, 2000).

ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human animals in research, and for publishing and presentations. The principles endorsed by the Society follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed from the Council for International Organizations of Medical Sciences (CIOMS). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and is publication or presentation.

Authors of manuscripts submitted for publication in a journal of the Acoustical Society of America or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievances Committee of the ASA.

APPROVAL BY APPROPRIATE GOVERNING AUTHORITY

The ASA requires all authors to abide by the principles of ethical research as a prerequisite for participation in Society-wide activities (e.g., publication of papers, presentations at meetings, etc.). Furthermore, the Society endorses the view that all research involving human and non-human vertebrate animals requires approval by the appropriate governing authority (e.g., institutional review board [IRB], or institutional animal care and use committee [IACUC], Health Insurance Portability and Accountability Act [HIPAA], or by other governing authorities used in many countries) and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, then the intent of the ASA Ethical Principles described in this document must be met. All research involving the use of human or non-human animals must have met the ASA Ethical Principles prior to the materials being submitted to the ASA for publication or presentation.

USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

Research involving the use of human subjects should have been approved by an existing appropriate governing authority (e.g., an institutional review board [IRB]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol that has been approved by the appropriate and responsible-governing body, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentially.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. The office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s) if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing With Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. The research involved the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well being both of human and non-human vertebrate animals often require the use of intact individuals representing a wide variety of species in experiments designed to address reasonable scientific questions. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Science (CIOMS) document: "International Guiding Principles for Biomedical Research Involving Animals1985"). Research involving the use of vertebrate animals should have been approved by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

The proper and humane treatment of vertebrate animals in research demands that investigators:

1. Acquired, cared for, used, interacted with, observed, and disposed of animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
2. Are knowledgeable of applicable research methods and are experienced in the care of laboratory animals, supervised all procedures involving animals, and assumed responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
3. Have insured that the current research is not repetitive of previously published work.
4. Should have used alternatives (e.g., mathematical models, computer simulations, etc.) when possible and reasonable.
5. Must have performed surgical procedures that were under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
6. Have ensured that all subordinates who use animals as a part of their employment or education received instruction in research methods and in the care, maintenance, and handling of the species that were used, commensurate with the nature of their role as a member of the research team.
7. Must have made all reasonable efforts to minimize the number of vertebrate animals used, the discomfort, the illness, and the pain of all animal subjects.
8. Must have made all reasonable efforts to minimize any harm to the environment necessary for the safety and well being of animals that were observed or may have been affective as part of a research study.
9. Must have made all reasonable efforts to have monitored and then mitigated any possible adverse affects to animals that were observed as a function of the experimental protocol.
10. Who have used a procedure subjecting animals to pain, stress, or privation may have done so only when an alternative procedure was unavailable; the goal was justified by its prospective scientific, educational, or applied value; and the protocol had been approved by an appropriate review board.
11. Proceeded rapidly to humanely terminate an animal's life when it was necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by an appropriate review board.

PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Plagiarism

Authors must not have presented portions of another's work or data as their own under any circumstances.

Publication Credit

Authors have taken responsibility and credit, including authorship credit, only for work they have actually performed or to which they have substantially contributed. Principal authorship and other publication credits accurately reflect the relative scientific or professional contributions of the individuals involved, regardless of their relative status. Mere possession of an institutional position, such as a department chair, does not justify authorship credit. Minor contributions to the research or to the writing of the paper should have been acknowledged appropriately, such as in footnotes or in an introductory statement.

Duplicate Publication of Data

Authors did not publish, as original data, findings that have been previously published. This does not preclude the republication of data when they are accompanied by proper acknowledgment as defined by the publication policies of the ASA.

Reporting Research Results

If authors discover significant errors in published data, reasonable steps must be made in as timely a manner as possible to rectify such errors. Errors can be rectified by a correction, retraction, erratum, or other appropriate publication means.

DISCLOSURE OF CONFLICTS OF INTEREST

If the publication or presentation of the work could directly benefit the author(s), especially financially, then the author(s) must disclose the nature of the conflict:

1) The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.

2) If the publication or presentation of the research would directly lead to the financial gain of the authors(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper.

3) If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Sustaining Members of the Acoustical Society of America

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: Elaine@acousticalsociety.org

Acentech Incorporated
www.acentech.com
Cambridge, Massachusetts
Consultants in Acoustics, Audiovisual and Vibration

ACO Pacific Inc.
www.acopacific.com
Belmont, California
Measurement Microphones, the ACOustic Interface™ System

American Institute of Physics
www.aip.org
College Park, Maryland
Career resources, undergraduate education, science policy, and history

Applied Physical Sciences Corp.
www.aphysci.com
Groton, Connecticut
Acoustics, Hydrodynamics, Electromagnetics, Manufacturing

BBN Technologies
www.bbn.com
Cambridge, Massachusetts
R&D company providing custom advanced research based solutions

Boeing Commercial Airplane Group
www.boeing.com
Seattle, Washington
Producer of Aircraft and Aerospace Products

D'Addario & Company, Inc.
www.daddario.com
Farmingdale, New York
D'Addario strings for musical instruments, Evans drumheads, Rico woodwind reeds and Planet Waves accessories

G.R.A.S.
Sound & Vibration ApS
www.gras.dk
Vedbaek, Denmark
Measurement microphones, Intensity probes, Calibrators

InfoComm International Standards
www.infocomm.org
Fairfax, Virginia
Advancing Audiovisual Communications Globally

International Business Machines Corporation
www.ibm.com/us
Yorktown Heights, New York
Manufacturer of Business Machines

Knowles Electronics, Inc.
www.knowles.com
Itasca, Illinois
Manufacturing Engineers: Microphones, Recording, and Special Audio Products

Massa Products Corporation
www.massa.com
Hingham, Massachusetts
Design and Manufacture of Sonar and Ultrasonic Transducers
Computer-Controlled OEM Systems

Meyer Sound Laboratories, Inc.
www.meyersound.com
Berkeley, California
Manufacture Loudspeakers and Acoustical Test Equipment

National Council of Acoustical Consultants
www.ncac.com
Indianapolis, Indiana
An Association of Independent Firms Consulting in Acoustics

Raytheon Company
Integrated Defense Systems
www.raytheon.com
Portsmouth, Rhode Island
Sonar Systems and Oceanographic Instrumentation: R&D in Underwater Sound Propagation and Signal Processing

Shure Incorporated
www.shure.com
Niles, Illinois
Design, development, and manufacture of cabled and wireless microphones for broadcasting, professional recording, sound reinforcement, mobile communications, and voice input–output applications; audio circuitry equipment; high fidelity phonograph cartridges and styli: automatic mixing systems; and related audio components and accessories. The firm was founded in 1925.

Thales Underwater Systems
www.thales-naval.com
Somerset, United Kingdom
Prime contract management, customer support services, sonar design and production, masts and communications systems design and production

3M Occupational Health & Environmental Safety Division
www.3m.com/occsafety
Minneapolis, Minnesota
Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
Hearing conservation resource center
www.e-a-r.com/hearingconservation

Wenger Corporation
www.wengercorp.com
Owatonna, Minnesota
Design and Manufacturing of Architectural Acoustical Products including Absorbers, Diffusers, Modular Sound Isolating Practice Rooms, Acoustical Shells and Clouds for Music Rehearsal and Performance Spaces

Wyle Laboratories
www.wyle.com
Arlington, Virginia
The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

Name of Company ______________________________

Address ______________________________

Telephone: ____________________ Fax: ____________________

E-mail: ____________________ WWW: ____________________

Size of Business: ☐ Small business ☐ Small business—No Journal ☐ Large business

Type of Business ______________________________

Please enclose a copy of your organization's brochure.

In listing of Sustaining Members in the Journal and on the ASA homepage we should like to indicate our products or services as follows:

____

(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

____

It is understood that a Sustaining Member will not use the membership for promotional purposes.

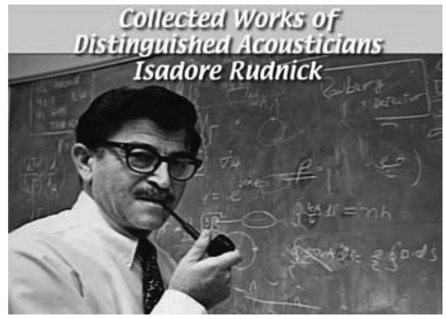
Signature of company representatives making application:

____

Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360.

Collected Works of Distinguished Acousticians

Isadore Rudnick



The first in this series of the Collected Works of Distinguished Acousticians is that of Isadore Rudnick (May 8, 1917 - August 22, 1997). Rudnick was honored by the Acoustical Society of America (ASA) with the R. Bruce Lindsay (Biennial) Award in 1948, the Silver Medal in Physical Acoustics in 1975, and the Gold Medal in 1982. He was recognized for his acoustics research in low temperature physics with this field's most prestigious award, the Fritz London Memorial Award, in 1981 and was inducted into the National Academy of Science in 1983. Izzy's research in physical acoustics addressed boundary propagation, reciprocity calibration, high intensity sound and its biological effects, nonlinear sound propagation, and acoustics in superconductors and superfluids, including critical phenomena in bulk and thin films. The first disc in this three disc set contains reprints of Rudnick's papers from scientific journals, including 26 from the Journal of the Acoustical Society of America, and 87 from other prestigious journals, as well as some consulting reports and invited papers presented at international meetings which would otherwise be difficult to obtain. The second disc includes a montage of photographs of Rudnick with colleagues and family, Rudnick's prize winning film "The Unusual Properties of Liquid Helium", and a video of the Plenary session at the ASA's 100th meeting where Rudnick presented 90 minutes of unique and stage-sized acoustics demonstrations. While videotaped under poor conditions and of lamentable quality, the reprocessed video of acoustics demonstrations is one of the most valuable parts of this collection. The third disc is a video recording of the Memorial Session held at the 135th meeting of the ASA, which provides a comprehensive summary of Rudnick's contributions as described by former students and collaborators.

The CD was compiled by Julian D. Maynard and Steven L. Garrett of the Pennsylvania State University, State College, Pennsylvania.

ORDER FORM

Price: $50-ASA members; $60-Nonmembers. Prepayment required by check or money order in U.S. dollars drawn on a bank in the U.S. or by Visa, MasterCard or American Express credit card.

Send orders to the Acoustical Society of America, Suite 1N01, 2 Huntington Quadrangle, Melville, NY 11747-4502, Tel.: 516-576-2360; Fax: 516-576-2377; Email: asa@aip.org.

Name____

Address____

City____________________State/Province__________Zip/Postal Code________Country____________

Email:____

Quantity	Price	ASA Member Number	Postage/Handling	Total
_______	@ $50.00/ea./Members	____________________	$5.00	$______
_______	@ $60.00/ea./Nonmembers		$5.00	$______
			Total Due and Enclosed	$_______

Credit Card #______________________________**Exp. Date**_________***Security Code**___________

Cardholder's Name____

Signature:____

***What is this?** On MasterCard and Visa cards, this is the 3-digit number printed in the signature area of the back of the card (last 3 digits AFTER the credit card number). On American Express cards it is the 4-digit card verification number on the front above the credit card number.

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please use our secure web page to process your credit card payment (http://www.abdi-ecommerce10.com/asa) or securely fax this form to (516-576-2377).

MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

BENEFITS OF MEMBERSHIP	Full Member	Associate	ce-Associate	Student
JASA Online–Vol. 1 (1929) to present	*	*	*	*
JASA tables of contents e-mail alerts	*	*	*	*
JASA, printed or CD ROM	*	*		
JASA Express Letters–online	*	*	*	*
Acoustics Today–the quarterly magazine	*	*	*	*
Proceedings of Meetings on Acoustics	*	*	*	*
Noise Control and *Sound*, *It's Uses and Control*–online archival magazines	*	*	*	*
Acoustics Research Letters Online (ARLO)–online archive	*	*	*	*
Programs for Meetings	Online	Online	Online	Online
Meeting Calls for Papers	Online	Online	Online	Online
Reduced Meeting Registration Fees	*	*		*
5 Free ASA standards per year-download only	*	*		*
Standards Discounts	*	*		*
Society Membership Directory	Online	Online	Online	Online
Electronic Announcements	*	*	*	*
Physics Today	*	*	*	*
Eligibility to vote and hold office in ASA	*			
Eligibility to be elected Fellow	*	*		
Participation in ASA Committees	*	*	*	*

QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

Student: Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $45 per year.
Associate: Any individual interested in acoustics. Dues: $95 per year. After five years, the dues of an Associate increase to that of a full Member.
Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $45 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $95 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- CD-ROM. The CD ROM mailed bimonthly. This option includes all of the material published in the Journal on CD ROM. **Cost: $35 in addition to dues.**
- COMBINATION OF THE CD-ROM AND PRINTED JOURNAL. The CD-ROM mailed bimonthly and the printed journal mailed monthly. **Cost: $70 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $165 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

ACOUSTICAL SOCIETY OF AMERICA

1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, asa@aip.org

For Office Use Only
Dues Rcvd ________
Aprvd by Ed ________
Aprvd by EC ________

APPLICATION FOR MEMBERSHIP

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT

☐ NON-MEMBER APPLYING FOR:
☐ MEMBER REQUESTING TRANSFER TO:

☐ STUDENT MEMBERSHIP
☐ ASSOCIATE MEMBERSHIP
☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP
☐ FULL MEMBERSHIP

Note that your choice of journal option *may* increase or decrease the amount you must remit.

SELECT JOURNAL OPTION:

Student members will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45. (Note: Student members may also receive the Journal on CD ROM at an additional charge of $35.)

Corresponding Electronic Associate Members will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $45.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$95
[ ] Online access plus print Journal $130
[ ] Online access plus CD ROM—$130
[ ] Online access plus print Journal and CD ROM combination—$165

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

OPTIONAL AIR DELIVERY: Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $165. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

LAST NAME	FIRST NAME	MIDDLE INITIAL	MS/MR/MRS/DR/PROF
HOME ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	MOBILE PHONE: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)			
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male		
HIGHEST ACADEMIC DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➛

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

☐ ACOUSTICAL OCEANOGRAPHY **M**
☐ ANIMAL BIOACOUSTICS **L**
☐ ARCHITECTURAL ACOUSTICS **A**
☐ BIOMEDICAL ACOUSTICS **K**
☐ ENGINEERING ACOUSTICS **B**
☐ MUSICAL ACOUSTICS **C**
☐ NOISE & NOISE CONTROL **D**
☐ PHYSICAL ACOUSTICS **E**
☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
☐ SIGNAL PROCESSING IN ACOUSTICS **N**
☐ SPEECH COMMUNICATION **H**
☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, addresses, and signatures of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
ADDRESS OF REFERENCE	ADDRESS OF SECOND REFERENCE
SIGNATURE OF REFERENCE (required for Full Member applications only)	SIGNATURE OF SECOND REFERENCE (required for Full Member applications only)
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300.

METHOD OF PAYMENT

☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

☐ American Express ☐ VISA ☐ MasterCard Signature ______________________________

(Credit card orders must be signed)

Account Number ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐

Expiration Date Mo. ☐☐ Yr. ☐☐

Security Code ☐☐☐☐

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please return this form by Fax (631-923-2875) or by postal mail.

Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn M. Hoglund, Ohio State University, Columbus, OH 43204, hoglund1@osu.edu and Sandra L. Guzman, Columbia College Chicago, Chicago, IL 60605, sguzman@colum.edu.

AUSTIN STUDENT CHAPTER
Benjamin C. Treweek
10000 Burnet Rd.
Austin, TX 78758
Email: btreweek@utexas.edu

BRIGHAM YOUNG UNIVERSITY STUDENT CHAPTER
Kent L. Gee
Dept. of Physics & Astronomy
Brigham Young Univ.
N283 ESC
Provo, UT 84602
Email: kentgee@byu.edu
www.acoustics.byu.edu

CHICAGO
Scott D. Pfeiffer
Threshold Acoustics LLC
141 W. Jackson Blvd.
Chicago, IL 60604
Email: spreiffer@thresholdacoustics.com

UNIVERSITY OF CINCINNATI STUDENT CHAPTER
Kyle T. Rich
Biomedical Engineering
Univ. of Cincinnati
231 Albert Sabin Way
Cincinnati, OH 45267
Email: richkt@mail.uc.edu

COLUMBIA COLLEGE CHICAGO STUDENT CHAPTER
Sandra Guzman
Dept. of Audio Arts and Acoustics
Columbia College Chicago
33 E. Congress Pkwy., Rm. 6010
Chicago, IL 60605
Email: sguzman@colum.edu

FLORIDA
Richard J. Morris
Communication Science and Disorders
Florida State Univ.
201 W. Bloxham
Tallahassee, FL 32306-1200
Email: richard.morris@cci.fsu.edu

GEORGIA INSTITUTE OF TECHNOLOGY STUDENT CHAPTER
Charlise Lemons
Georgia Institute of Technology
Atlanta, GA 30332-0405
Email: clemons@gatech.edu

GREATER BOSTON
Eric Reuter
Reuter Associates, LLC
10 Vaughan Mall, Ste. 201A
Portsmouth, NH 03801
Email: ereuter@reuterassociates.com

UNIVERSITY OF HARTFORD STUDENT CHAPTER
Robert Celmer
Mechanical Engineering Dept., UT-205
Univ. of Hartford
200 Bloomfield Ave.
West Hartford, CT 06117
Email: celmer@hartford.edu

UNIVERSITY OF KANSAS STUDENT CHAPTER
Robert C. Coffeen
Univ. of Kansas
School of Architecture, Design, and Planning
Marvin Hall
1465 Jayhawk Blvd.
Lawrence, KS 66045
Email: coffeen@ku.edu

LOS ANGELES
Neil A. Shaw
www.asala.org

MID-SOUTH
Tiffany Gray
NCPA
Univ. of Mississippi
University, MS 38677
Email: midsouthASAchapter@gmail.com

NARRAGANSETT
David A. Brown
Univ. of Massachusetts, Dartmouth
151 Martime St.
Fall River, MA 02723
Email: dbacoustics@cox.net

UNIVERSITY OF NEBRASKA STUDENT CHAPTER
Hyun Hong
Architectural Engineering
Univ. of Nebraska
Peter Kiewit Institute
1110 S. 67th St.
Omaha, NE 68182-0681
Email: unoasa@gmail.com

NORTH CAROLINA
Noral Stewart
Stewart Acoustical Consultants
7330 Chapel Hill Rd., Ste.101
Rayleigh, NC
Email: noral@sacnc.com

NORTH TEXAS
Peter F. Assmann
School of Behavioral and Brain Sciences
Univ. of Texas-Dallas
Box 830688 GR 4.1
Richardson, TX 75083
Email: assmann@utdallas.edu

NORTHEASTERN UNIVERSITY STUDENT CHAPTER
Victoria Suha
Email: suha.v@husky.neu.ed

OHIO STATE UNIVERSITY STUDENT CHAPTER
Kayla Karg
The Ohio State Univ.
Columbus, OH 43210
Email: karg.24@osu.edu

PENNSYLVANIA STATE UNIVERSITY STUDENT CHAPTER
Anand Swaminathan
Pennsylvania State Univ.
201 Applied Science Bldg.
University Park, PA 16802
Email: azs563@psu.edu
www.psuasa.org

PHILADELPHIA
Kenneth W. Good, Jr.
Armstrong World Industries, Inc.
2500 Columbia Ave.
Lancaster, PA 17603
Email: kwgoodjr@armstrong.com

PURDUE UNIVERSITY STUDENT CHAPTER
Kao Ming Li
Purdue Univ.
585 Purdue Mall
West Lafayette, IN 47907
Email: mmkmli@purdue.edu
Email: purdueASA@gmail.com

SEATTLE STUDENT CHAPTER

Camilo Perez
Applied Physics Lab.
Univ. of Washington
1013 N.E. 40th St,
Seattle, WA 98105-6698
Email: camipiri@uw.edu

UPPER MIDWEST

David Braslau
David Braslau Associates, Inc.
6603 Queen Ave. South, Ste. N
Richfield, MN 55423
Email: david@braslau.com

WASHINGTON, DC

Shane Guan
National Marine Fisheries Service
Office of Protected Resources
1315 East-West Hwy., Ste. 13826
Silver Spring, MD 20910
Email: Shane.guan@noaa.gov

ACOUSTICAL SOCIETY OF AMERICA • BOOKS, CDS, DVD, VIDEOS ON ACOUSTICS
Order online at http://www.abdi-ecommerce10.com/asa

ACOUSTICAL DESIGN OF MUSIC EDUCATION FACILITIES. Edward R. McCue and Richard H. Talaske, Eds. Plans, photographs, and descriptions of 50 facilities with explanatory text and essays on the design process. 236 pp, paper, 1990. Price: $23. **Item # 0-88318-8104**

ACOUSTICAL DESIGN OF THEATERS FOR DRAMA PERFORMANCE: 1985--2010. David T. Bradley, Erica E. Ryherd, & Michelle C. Vigeant, Eds. Descriptions, color images, and technical and acoustical data of 130 drama theatres from around the world, with an acoustics overview, glossary, and essays reflecting on the theatre design process. 334 pp, hardcover 2010. Price: $45. **Item #978-0-9846084-5-4**

ACOUSTICAL DESIGNING IN ARCHITECTURE. Vern O. Knudsen and Cyril M. Harris. Comprehensive, non-mathematical treatment of architectural acoustics; general principles of acoustical designing. 408 pp, paper, 1980 (original published 1950). Price: $23. **Item # 0-88318-267X**

ACOUSTICAL MEASUREMENTS. Leo L. Beranek. Classic text with more than half revised or rewritten. 841 pp, hardcover 1989 (original published 1948). **Available on Amazon.com**

ACOUSTICS. Leo L. Beranek. Source of practical acoustical concepts and theory, with information on microphones, loud-speakers and speaker enclosures, and room acoustics. 491 pp, hardcover 1986 (original published 1954). **OUT-OF-PRINT**

ACOUSTICS—AN INTRODUCTION TO ITS PHYSICAL PRINCIPLES AND APPLICATIONS. Allan D. Pierce. Textbook introducing the physical principles and theoretical basis of acoustics, concentrating on concepts and points of view that have proven useful in applications such as noise control, underwater sound, architectural acoustics, audio engineering, nondestructive testing, remote sensing, and medical ultrasonics. Includes problems and answers. 678 pp, hardcover 1989 (original published 1981). Price: $33. **Item # 0-88318-6128**

ACOUSTICS, ELASTICITY AND THERMODYNAMICS OF POROUS MEDIA: TWENTY-ONE PAPERS BY M. A. BIOT. Ivan Tolstoy, Ed. Presents Biot's theory of porous media with applications to acoustic wave propagation, geophysics, seis-mology, soil mechanics, strength of porous materials, and visco-elasticity. 272 pp, hardcover 1991. Price: $28. **Item # 1-56396-0141**

ACOUSTICS OF AUDITORIUMS IN PUBLIC BUILDINGS, Leonid I. Makrinenko, John S. Bradley, Ed. Presents developments resulting from studies of building physics. 172 pp, hardcover 1994 (original published 1986) Price: $38. **Item # 1-56396-3604**

ACOUSTICS OF WORSHIP SPACES, David Lubman and Ewart A. Wetherill, Eds. Drawings, photographs, and accompanying data of worship houses provide information on the acoustical design of chapels, churches, mosques, temples, and synagogues. 91 pp, paper 1985. Price: $23. **Item # 0-88318-4664**. **OUT-OF-PRINT**

ASA EDITION OF SPEECH AND HEARING IN COMMUNICATION. Harvey Fletcher; Jont B. Allen, Ed. A summary of Harvey Fletcher's 33 years of acoustics work at Bell Labs. A new introduction, index, and complete bibliography of Fletcher's work are important additions to this classic volume. 487 pp, hardcover 1995 (original published 1953). Price: $40. **Item # 1-56396-3930**

AEROACOUSTICS OF FLIGHT VEHICLES: THEORY AND PRACTICE. Harvey H. Hubbard, Ed. Two-volumes oriented toward flight vehicles emphasizing the underlying concepts of noise generation, propagation, predicting and control. Vol. 1 589 pp/Vol. 2 426 pp, hardcover 1994 (original published 1991). Price per 2-vol. set: $58. **Item # 1-56396-404X**

COLLECTED PAPERS ON ACOUSTICS. Wallace Clement Sabine. Classic work on acoustics for architects and acousticians. 304 pp, hardcover 1993 (originally published 1921). Price: $28. **Item # 0-932146-600**

CONCERT HALLS AND OPERA HOUSES. Leo L. Beranek. Over 200 photos and architectural drawings of 100 concert halls and opera houses in 31 countries with rank-ordering of 79 halls and houses according to acoustical quality. 653 pp. hardcover 2003. Price: $50. **Item # 0-387-95524-0**. **OUT-OF-PRINT**

CRYSTAL ACOUSTICS. M.J.P. Musgrave. For physicists and engineers who study stress-wave propagation in anisotropic media and crystals. 406 pp. hardcover (originally published 1970). Price: $34. **Item # 0-9744067-0-8**

DEAF ARCHITECTS & BLIND ACOUSTICIANS? Robert E. Apfel. A primer for the student, the architect and the planner. 105 pp. paper 1998. Price: $22. **Item #0-9663331-0-1**

THE EAR AS A COMMUNICATION RECEIVER. Eberhard Zwicker & Richard Feldtkeller. Translated by Hannes Müsch, Søren Buus, Mary Florentine. Translation of the classic *Das Ohr Als Nachrichtenempfänger*. Aimed at communication engineers and sensory psychologists. Comprehensive coverage of the excitation pattern model and loudness calculation schemes. 297 pp, hardcover 1999 (original published 1967). Price: $50. **Item # 1-56396-881-9**

ELECTROACOUSTICS: THE ANALYSIS OF TRANSDUCTION, AND ITS HISTORICAL BACK-GROUND. Frederick V. Hunt. Analysis of the conceptual development of electroacoustics including origins of echo ranging, the crystal oscillator, evolution of the dynamic loudspeaker, and electromechanical coupling, 260 pp, paper 1982 (original published 1954). **Available on Amazon.com**

ELEMENTS OF ACOUSTICS. Samuel Temkin. Treatment of acoustics as a branch of fluid mechanics. Main topics include propagation in uniform fluids at rest, trans-mission and reflection phenomena, attenuation and dispersion, and emission. 515 pp. hardcover 2001 (original published 1981). Price: $30. **Item #1-56396-997-1**

EXPERIMENTS IN HEARING. Georg von Békésy. Classic on hearing containing vital roots of contemporary auditory knowledge. 760 pp, paper 1989 (original published 1960). Price: $23. **Item #0-88318-6306**

FOUNDATIONS OF ACOUSTICS. Eugen Skudrzyk. An advanced treatment of the mathematical and physical foundations of acoustics. Topics include integral transforms and Fourier analysis, signal processing, probability and statistics, solutions to the wave equation, radiation and diffraction of sound. 790 pp. hardcover 2008 (originally published 1971). Price: $60. **Item # 3-211-80988-0**

HALLS FOR MUSIC PERFORMANCE: TWO DECADES OF EXPERIENCE, 1962–1982. Richard H. Talaske, Ewart A. Wetherill, and William J. Cavanaugh, Eds. Drawings, photos, and technical and physical data on 80 halls; examines standards of quality and technical capabilities of performing arts facilities. 192 pp, paper 1982. Price: $23. **Item # 0-88318-4125**

HALLS FOR MUSIC PERFORMANCE: ANOTHER TWO DECADES OF EXPERIENCE 1982–2002. Ian Hoffman, Christopher Storch, and Timothy Foulkes, Eds. Drawings, color photos, technical and physical data on 142 halls. 301 pp, hardcover 2003. Price: $56. **Item # 0-9744067-2-4**

HANDBOOK OF ACOUSTICAL MEASUREMENTS AND NOISE CONTROL, THIRD EDITION. Cyril M. Harris. Comprehensive coverage of noise control and measuring instruments containing over 50 chapters written by top experts in the field. 1024 pp, hardcover 1998 (original published 1991). **OUT-OF-PRINT**

HEARING: ITS PSYCHOLOGY AND PHYSIOLOGY. Stanley Smith Stevens & Hallowell Davis. Volume leads readers from the fundamentals of the psycho-physiology of hearing to a complete understanding of the anatomy and physiology of the ear. 512 pp, paper 1983 (originally published 1938). **OUT-OF-PRINT**

NONLINEAR ACOUSTICS, Mark F. Hamilton and David T. Blackstock. Research monograph and reference for scientists and engineers, and textbook for a graduate course in nonlinear acoustics. 15 chapters written by leading experts in the field. 455 pp, hardcover, 2008 (originally published in 1996). Price: $45. **Item # 0-97440-6759**

NONLINEAR ACOUSTICS. Robert T. Beyer. A concise overview of the depth and breadth of nonlinear acoustics with an appendix containing references to new developments.452 pp., hardcover, 1997 (originally published 1974). Price: $45. **Item # 1-56396-724-3**

NONLINEAR UNDERWATER ACOUSTICS. B. K. Novikov, O. V. Rudenko, V. I. Timoshenko. Translated by Robert T. Beyer. Applies the basic theory of nonlinear acoustic propagation to directional sound sources and receivers, including design nomographs and construction details of parametric arrays. 272 pp, paper 1987. Price: $34. **Item # 0-88318-5229**

OCEAN ACOUSTICS. Ivan Tolstoy and Clarence S. Clay. Presents the theory of sound propagation in the ocean and compares the theoretical predictions with experimental data. Updated with reprints of papers by the authors supplementing and clarifying the material in the original edition. 381 pp, paper 1987 (original published 1966). **OUT-OF-PRINT**

ORIGINS IN ACOUSTICS, Frederick V. Hunt. History of acoustics from antiquity to the time of Isaac Newton. 224 pp, hardcover 1992. Price: $19. **Item # 0-300-022204**.
OUT-OF-PRINT

PAPERS IN SPEECH COMMUNICATION. Papers charting four decades of progress in understanding the nature of human speech production, and in applying this knowledge to problems of speech processing. Contains papers from a wide range of journals from such fields as engineering, physics, psychology, and speech and hearing science. 1991, hardcover.

Speech Perception, Joanne L. Miller, Raymond D. Kent, Bishnu S. Atal, Eds. 764 pp. **OUT-OF-PRINT**
Speech Production, Raymond D. Kent, Bishnu S. Atal, Joanne L. Miller, Eds. 880 pp. **OUT-OF-PRINT**
Speech Processing, Bishnu S. Atal, Raymond D. Kent, Joanne L. Miller, Eds. 672 pp. Price: $38. **Item # 0-88318-9607**

PROPAGATION OF SOUND IN THE OCEAN. Contains papers on explosive sounds in shallow water and long-range sound transmission by J. Lamar Worzel, C. L. Pekeris, and Maurice Ewing. hardcover 2000 (original published 1948). Price: $37. **Item #1-56396-9688**

RESEARCH PAPERS IN VIOLIN ACOUSTICS 1975–1993. Carleen M. Hutchins, Ed., Virginia Benade, Assoc. Ed. Contains 120 research papers with an annotated bibliography of over 400 references. Introductory essay relates the development of the violin to the scientific advances from the early 15th Century to the present. Vol. 1, 656 pp; Vol. 2, 656 pp. hardcover 1996. Price: $120 for the two-volume set. **Item # 1-56396-6093**

RIDING THE WAVES. Leo L. Beranek. A life in sound, science, and industry. 312 pp. hardcover 2008. **OUT-OF-PRINT**

THE SABINES AT RIVERBANK. John W. Kopec. History of Riverbank Laboratories and the role of the Sabines (Wallace Clement, Paul Earls, and Hale Johnson) in the science of architectural acoustics. 210 pp. hardcover 1997. Price:$19. **Item # 0-932146-61-9**

SONICS, TECHNIQUES FOR THE USE OF SOUND AND ULTRASOUND IN ENGINEERING AND SCIENCE. Theodor F. Hueter and Richard H. Bolt. Work encompassing the analysis, testing, and processing of materials and products by the use of mechanical vibratory energy. 456 pp, hardcover 2000 (original published 1954). Price: $30. **Item # 1-56396-9556**

SOUND IDEAS, Deborah Melone and Eric W. Wood. Early days of Bolt Beranek and Newman Inc. to the rise of Acentech Inc. 363 pp. hardcover 2005. Price: $25. **Item # 200-692-0681**

SOUND, STRUCTURES, AND THEIR INTERACTION, Miguel C. Junger and David Feit. Theoretical acoustics, structural vibrations, and interaction of elastic structures with an ambient acoustic medium. 451 pp, hardcover 1993 (original published 1972). Price: $23. **Item # 0-262-100347**

THEATRES FOR DRAMA PERFORMANCE: RECENT EXPERIENCE IN ACOUSTICAL DESIGN, Richard H. Talaske and Richard E. Boner, Eds. Plans, photos, and descriptions of theatre designs, supplemented by essays on theatre design and an extensive bibliography. 167 pp, paper 1987. Price: $23. **Item # 0-88318-5164**

THERMOACOUSTICS, Gregory W. Swift. A unifying thermoacoustic perspective to heat engines and refrigerators. 300 pp, paper, 2002. Price: $50. **Item # 0-7354-0065-2**

VIBRATION AND SOUND, Philip M. Morse. Covers the broad spectrum of acoustics theory, including wave motion, radiation problems, propagation of sound waves, and transient phenomena. 468 pp, hardcover 1981 (originally published 1936). Price: $28. **Item # 0-88318-2874**

VIBRATION OF PLATES, Arthur W. Leissa. 353 pp, hardcover 1993 (original published 1969). **OUT-OF-PRINT**

VIBRATION OF SHELLS, Arthur W. Leissa. 428 pp, hardcover 1993 (original published 1973). **Item # 1-56396-2934**

SET ITEM # 1-56396-KIT
Monographs dedicated to the organization and summarization of knowledge existing in the field of continuum vibrations. Price: $28; $50 for 2-volume set.

CDs, DVD, VIDEOS, STANDARDS

Auditory Demonstrations (CD). Teaching adjunct for lectures or courses on hearing and auditory effects. Provides signals for teaching laboratories. Contains 39 sections demonstrating various characteristics of hearing. Includes booklet containing introductions and narrations of each topic and bibliographies for additional information. Issued in1989. Price: $23. **Item # AD-CD-BK**

Measuring Speech Production (DVD). Demonstrations for use in teaching courses on speech acoustics, physiology, and instrumentation. Includes booklet describing the demonstrations and bibliographies for more information. Issued 1993. Price: $52. **Item # MS-DVD**

Scientific Papers of Lord Rayleigh (CD ROM). Over 440 papers covering topics on sounds, mathematics, general mechanics, hydrodynamics, optics and properties of gasses by Lord Rayleigh (John William Strutt) the author of the *Theory of Sound*. Price: $40. **Item #0-9744067-4-0**

Proceedings of the Sabine Centennial Symposium (CD ROM). Held June 1994. Price: $50. **Item # INCE25-CD**

Fifty Years of Speech Communication (VHS). Lectures presented by distinguished researchers at the ASA/ICA meeting in June 1998 covering development of the field of Speech Communication. Lecturers: G. Fant, K.N. Stevens, J.L. Flanagan, A.M. Liberman, L.A. Chistovich--presented by R.J. Porter, Jr., K.S. Harris, P. Ladefoged, and V. Fromkin. Issued in 2000. Price: $30. **Item # VID-Halfcent**

Speech Perception (VHS). Presented by Patricia K. Kuhl. Segments include: I. General introduction to speech/language processing; Spoken language processing; II. Classic issues in speech perception; III. Phonetic perception; IV. Model of developmental speech perception; V. Cross-modal speech perception: Links to production; VI. Biology and neuroscience connections. Issued 1997. Price: $30. **Item # SP-VID**

Standards on Acoustics. Visit the ASA Store (https://global.ihs.com/home_page_asa.cfm?&rid=ASA) to purchase or download National (ANSI) and International (ISO) Standards on topics ranging from measuring environmental sound to standards for calibrating microphones .

Order the following from ASA, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@aip.org; Fax: 631-923-2875 Telephone orders not accepted. Prepayment required by check (drawn on US bank) or by VISA, MasterCard, American Express.

Study of Speech and Hearing at Bell Telephone Laboratories (CD). Nearly 10,000 pages of internal documents from AT&T archives including historical documents, correspondence files, and laboratory notebooks on topics from equipment requisitions to discussions of project plans, and experimental results. Price: $20 (postage included).

Collected Works of Distinguished Acousticians CD - Isadore Rudnick (CD + DVD). 3 disc set includes reprints of papers by Isadore Rudnick from scientific journals, a montage of photographs with colleagues and family, and video recordings of the Memorial Session held at the 135th meeting of the ASA. Price $50 (postage included).

Technical Memoranda issued by Acoustics Research Laboratory-Harvard University (CD). The Harvard Research Laboratory was established in 1946 to support basic research in acoustics. Includes 61 reports issued between 1946 and 1971 on topics such as radiation, propagation, scattering, bubbles, cavitation, and properties of solids, liquids, and gasses. Price $25.00 (postage included).

ORDER FORM FOR ASA BOOKS, CDS, DVD, VIDEOS

Place your order online at http://www.abdi-ecommerce10.com/asa/ for faster processing.

1. Payment must accompany order. Payment may be made by check or international money order in U.S. funds drawn on U.S. bank or by VISA, MasterCard, or American Express credit card.

2. Send orders to: Acoustical Society of America, Publications, P. O. Box 1020, Sewickley, PA 15143-9998; Tel.: 412-741-1979; Fax: 412-741-0609.

3. A 10% discount applies on orders of 5 or more copies of the same title only.

4. Returns are not accepted.

Item #	Quantity	Title	Price	Total
		Subtotal		
		Shipping Costs (all orders) based on weight and distance. For quote call 412-741-1979, visit http://www.abdi-ecommerce10.com/asa, or email asapubs@abdintl.com		
		10% discount of orders of 5 or more of the same title		
		Total		

Name____

Address____

____

City_________________________ State______ ZIP _________Country______________

Tel.: __________________Fax: __________________Email:______________________

Method of Payment

[ ] Check or money order enclosed for $__________ (U.S. funds/drawn on U.S. bank made payable to the Acoustical Society of America)

[ ] American Express [ ] VISA [ ] MasterCard

Cardholders signature____

(Credit card orders must be signed)

Account # ____________________________________ Expires Mo.___________ Yr.___________

Due to security risks and Payment Card Industry (PCI) data security standards email is NOT an acceptable way to transmit credit card information. Please use our secure webpage to process your credit card payment (http://www.abdi-ecommerce10.com/asa/) or securely fax this form to (631-923-2875).

AUTHOR INDEX

to papers presented at

170th Meeting: Acoustical Society of America

INDEX TO ADVERTISERS

ADVERTISING SALES OFFICE

JOURNAL ADVERTISING SALES
Robert G. Finnegan, Director, Journal Advertising
AIP Publishing, LLC
1305 Walt Whitman Road, Suite 300
Melville, NY 11747-4300
Telephone: 516-576-2433
Fax: 516-576-2481
Email: rfinnegan@aip.org

SR. ADVERTISING PRODUCTION MANAGER
Christine DiPasca
Telephone: 516-576-2434
Fax: 516-576-2481
Email: cdipasca@aip.org