The Journal of the Acoustical Society of America

Vol. 142, No. 4, Pt. 2 of 2, October 2017 www.acousticalsociety.org

174th Meeting
Acoustical Society of America

New Orleans Marriott Hotel
New Orleans, Louisiana
4–8 December 2017

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The Journal of the Acoustical Society of America

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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 Director of Operations, 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

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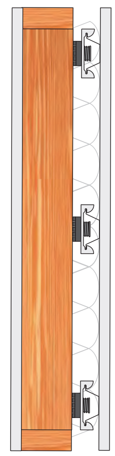

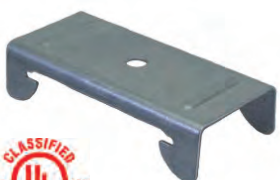

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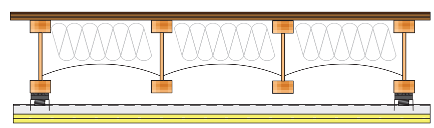

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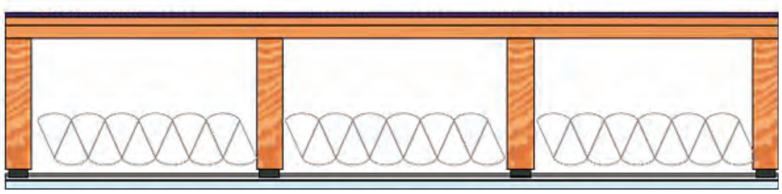

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TECHNICAL PROGRAM SUMMARY
***Indicates Special Session**

MONDAY MORNING

*1aAA Impact of Entertainment Sound on Communities
*1aAO Oceanographic Contributions to the Characteristics and Variability of the Underwater Soundscape
1aEAa General Topics in Engineering Acoustics I
1aEAb General Topics in Engineering Acoustics II
*1aNS Supersonic Jet and Rocket Noise I
*1aSA Standards in Structural Acoustics and Vibration
*1aSP Source Tracking with Microphone/Hydrophone Arrays I
1aUW Underwater Acoustic Scattering and Reverberation

MONDAY AFTERNOON

*1pAA Perceived Diffuseness I
*1pAB Bioacoustic Contributions to the Characteristics and Variability of Soundscapes, Underwater or Terrestrial
*1pAO Acoustic Scattering from Hydrocarbons and Hydrothermal Vents
1pEAa General Topics in Engineering Acoustics III
1pEAb General Topics Engineering Acoustics IV
*1pSP Source Tracking with Microphone/Hydrophone Arrays II
1pUW Underwater Acoustic Propagation: Models, Methods, and Statistics

MONDAY EVENING

*1eIDa Special Presentation on The Clarinet in Early New Orleans Jazz
*1eIDb Tutorial Lecture on Infrasound Phenomenology, Propagation, and Detection

TUESDAY MORNING

*2aAA Performance Spaces for Modern Music
*2aAB In Memory of George Ioup: Acoustics in the Gulf of Mexico I
*2aBA Wave Propagation in Complex Media: From Theory to Applications I
*2aEA Thermophone Transduction
*2aED Undergraduate Research Exposition (Poster Session)
*2aMU Measurement Methods and Instrumentation for Musical Acoustics
*2aPA Sound Used as an Investigative Tool for Industrial Solutions
*2aSA Acoustic Metamaterials I
*2aSC Articulatory and Acoustic Characteristics of Nasalization
*2aSP Detection, Classification, Localization and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) I
*2aUW Sediment Characterization Using Direct and Inverse Techniques I

TUESDAY AFTERNOON

*2pAA Perceived Diffuseness II
*2pAB In Memory of George Ioup: Acoustics in the Gulf of Mexico II
*2pBA Wave Propagation in Complex Media: From Theory to Applications II
2pEA General Topics in Engineering Acoustics V
*2pEDa Acoustics Education Prize Lecture
2pEDb General Topics in Acoustics Education
*2pID Guidance from the Experts: Applying for Grants and Fellowships
*2pMUa Cajun Music: Accordians, Culture, and History
2pMUb Cajun Music Concert by the Savoy Family Bank
*2pNSa Evaluation of Acoustics in Hospitals and Healthcare Facilities
*2pNSb Acoustics and its Role in Accessibility (e.g. Americans with Disabilities Act)
2pPA General Topics in Physical Acoustics I
*2pSA Acoustic Metamaterials II
2pSC Speech Production (Poster Session)
*2pSP Detection, Classification, Localization and Tracking (DCLT) using Acoustics (and Perhaps Other Sensing Modalities) II
*2pUW Sediment Characterization Using Direct and Inverse Techniques II

WEDNESDAY MORNING

*3aAA Restaurant Acoustics
3aAB General Bioacoustics
*3aAO Munk Award Lecture
*3aBA Wave Propagation in Complex Media: From Theory to Applications III
*3aED Hands-On Acoustics Demonstrations for Middle- and High-School Students
*3aID Standards: Practical Applications in Acoustics
3aMU General Topics in Musical Acoustics
*3aPA Acoustofluidics
3aPPa Exploring the Perception of Sound (Poster Session)
3aPPb Perception and Physiology: Musicians, Musical Instruments, and the Body (Poster Session)
*3aSA Structural Acoustics and Vibration Applications of Finite Element Analysis, Boundary Element Analysis, and Statistical Element Analysis Computational Methods
*3aSC Teaching Phonetics and Speech Science in the New Millennium: Challenges and Opportunities
*3aSP Detection, Classification, Localization, and Tracking (DCLT) using Acoustics (and Perhaps Other Sensing Modalities) III
*3aUWa Sediment Characterization Using Direct and Inverse Techniques III
*3aUWb Session in Honor of Chester McKinney

WEDNESDAY AFTERNOON

*3pAA Speech Privacy Concerns in Open Plan Spaces
3pBA Imaging I
*3pED Listen Up and Get Involved
*3pIDa Hot Topics: Hunt is Still Hot
*3pIDb ASA Hunt Postdoctoral Research Fellows: Through the Years (Poster Session)
*3pNS Urban Planning Using Soundscape I
3pPA General Topics in Physical Acoustics II
3pSC Clinical Populations (Poster Session)
*3pSP Detection, Classification, Localization, and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) IV

THURSDAY MORNING

*4aAA Speech Intelligibility in Reverberation and Noise
4aAB General Biosonar
*4aAO Biological Effects on Seabed Geoacoustic Properties I
*4aBA Ultrasound-Mediated Neuromodulation
*4aNS Urban Planning using Soundscape II
4aPA Infrasound, Atmospheric Sound Propagation, and Turbulence
4aPP Psychoacoustics of Speech Perception in Noise, Localization, and Frequency Selectivity
*4aSCa The Southern States: Social Factors and Language Variation I
*4aSCb The Southern States: Social Factors and Language Variation II (Poster Session)
*4aSP Signal Processing in Acoustics Metamaterials
4aUW Underwater Soundscapes and Noise: Measurement and Abatement

THURSDAY AFTERNOON

*4pAA Back to the Future: A Look at Multipurpose Spaces, How They've Changed & What's Next
*4pAB Neurophysiology of Echolocation
*4pAO Biological Effects on Seabed Geoacoustic Properties II
4pBA Imaging II
*4pED Synthetic Aperture Sonar for Youngsters
*4pNS Wind Turbine Noise
*4pPA Acoustics of Detecting Gravitational Waves using LIGO
4pSC Speech Perception and Word Recognition (Poster Session)
*4pSPa Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity I
*4pSPb Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity II
4pUW Arctic Acoustics

FRIDAY MORNING

5aAA Topics in Architectural Acoustics: Measurements, Modeling, and Isolation
5aAO Topics in Acoustical Oceanography
5aBA Therapeutic Ultrasound
*5aPA Nonlinear Elasticity in Geomaterials
5aSC Bilingual and Non-Native Speech Perception and Production (Poster Session)
5aSP Topics in Acoustic Signal Processing
5aUW Underwater Measurements and Applications

APPLICATIONS DUE 12 FEBRUARY 2018

PHYSICAL ACOUSTICS SUMMER SCHOOL

PASS
2018

The 2018 Physical Acoustics Summer School, PASS 2018, will be held at The Inn at Ole Miss Hotel & Conference Center on the Oxford campus of The University of Mississippi, with arrival on 03 June and departure on 08 June 2018. This is a change of venue from the prior two Physical Acoustics Summer Schools. 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. Oxford is about 80 miles from Memphis International Airport (MEM). Some free van transportation will be available. There is a $200 Student Registration Fee. The basic costs of participation, including room and board, based on standard multiple-occupancy for Students, will be paid by sponsors.

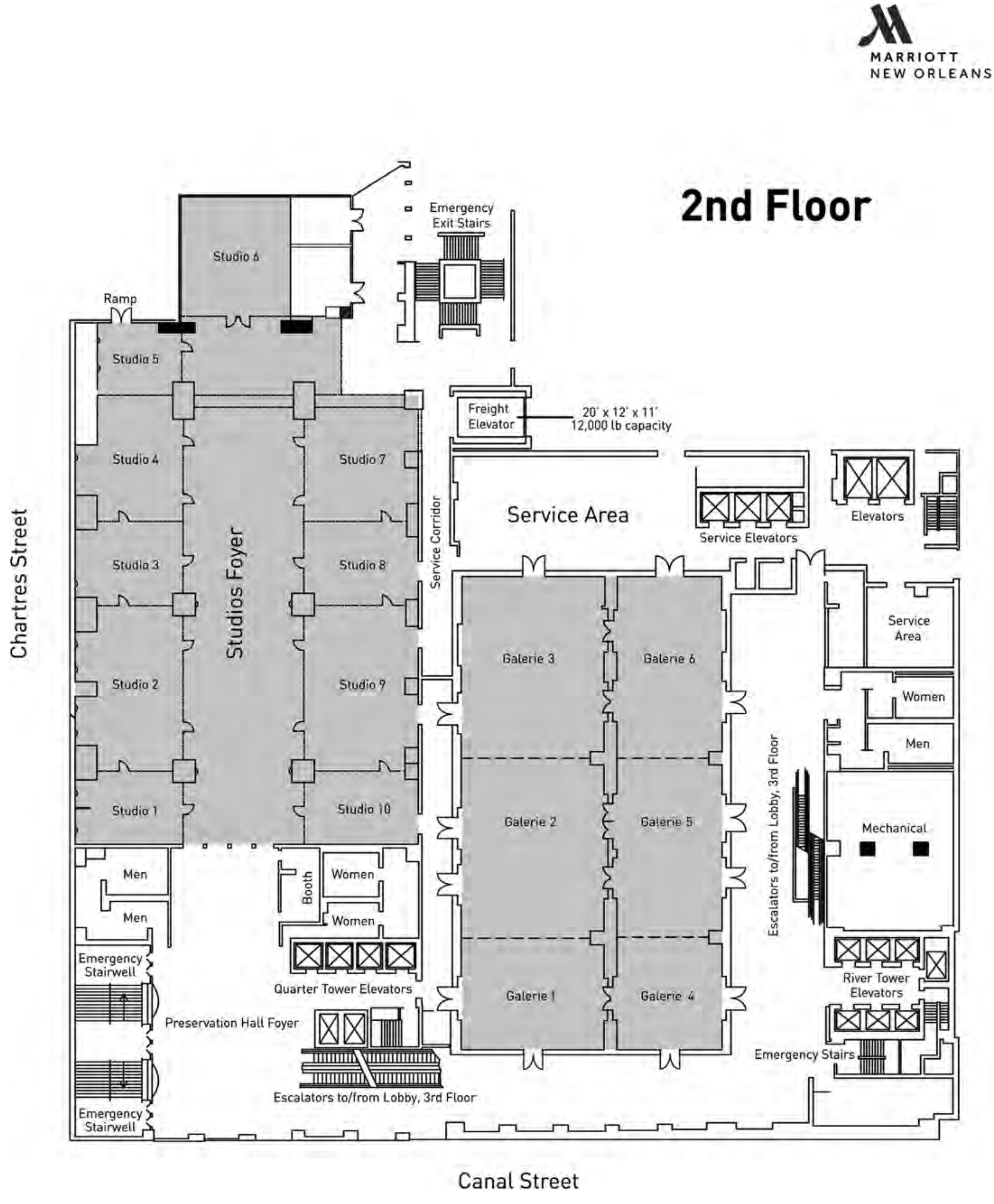
PROGRAM: Program information, including a full Preliminary Schedule, will be available to all those who request the Application Forms from the web page (below) at the National Center for Physical Acoustics (NCPA) at The University of Mississippi.

APPLICATION DEADLINE: Complete applications for the 2018 Physical Acoustics Summer School must be received at NCPA no later than Monday 12 February 2018.

APPLICATIONS: All participants must have a completed Application Form on file. Students must also provide a transcript and one professional reference letter. Copies of the Announcement, Application Form, and Preliminary Schedule will be available at http://ncpa.olemiss.edu/pass-2018/. The initial contact is Debra A. Bos, 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. Dr. J. R. (Josh) Gladden at Ole Miss is the PASS 2018 Director.

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

	Monday AM	Monday PM	Monday Eve	Tuesday AM	Tuesday PM	Tuesday Eve	Wednesday AM	Wednesday PM	Wednesday Eve	Thursday AM	Thursday PM	Thursday Eve	Friday AM
Acadia		1pSC 2:00		2aSC 9:00	2pSC 1:00		3aSC 8:00	3pSC 1:00					
Balcony I/J/K		1pPA 1:25			2pEDa 2:00 2pEDb 3:15		3aED 8:30		3eED 5:00		4pED 4:00		
Balcony L	1aSA 8:55	1pSA 3:30		2aPA 9:00	2pPA 1:00	TCPA 7:30	3aPA 8:30	3pPA 1:00		4aPA 9:00	4pPA 2:30		5aPA 8:00
Balcony M		1pAO 1:00		2aBA 9:25	2pBA 1:00	TCPP 7:30	3aBA 8:25	3pBA 1:00	TCBA 7:30	4aBA 8:25	4pBA 1:00		5aBA 9:00
Balcony N	1aEAa 9:00 1aEAb 10:30	1pEAa 1:00 1pEAb 2:45		2aEA 9:00	2pEA 1:00		3aSA 8:30						
Salon A/B/C		1pUW 1:00		2aSA 8:30	2pSA 1:00	TCAO 7:30	3aUWb 8:00			4aAO 8:00	4pAO 1:00	TCSC 7:30	5aAO 9:00
Salon D	1aSP 9:00	1pSP 1:00	1eIDb 7:00	2aSP 7:55	2pSP 1:00		3aSP 9:00	3pSP 1:00	TCSP 7:30	4aSP 8:00	4pSPa 1:00 4pSPb 4:05		5aSP 8:00
Salon E	1aUW 8:15		1eIDa 5:15	2aAB 8:35	2pAB 1:00 2pMUb 5:00		3aUWa 7:40 3aAO 11:00	3pIDa 1:00		4aUW 8:00	4pUW 1:00		5aUW 8:30
Salon F/G/H	1aAO 8:00	1pAB 1:00		2aUW 8:20	2pUW 1:00	TCAB 7:30	3aAB 8:30			4aAB 8:00	4pAB 1:00	TCUW 7:30	
Studio 2	1aNS 7:50	1pNS 1:00			2pNSa 1:00 2pNSb 3:30			3pNS 1:15		4aNS 7:55	4pNS 1:00	TCNS 7:30	
Studio 4		1pMU 1:00		2aMU 8:15	2pMUa 1:55	TCMU 7:30	3aMU 8:45			4aPP 9:00			
Studio 7					2pID 2:00	TCEA 4:30 TCSA 7:30	3aID 7:40			4aSCa 8:00			
Studio 9	1aAA 8:15	1pAA 1:00		2aAA 8:00	2pAA 1:15	TCAA 7:30	3aAA 7:45	3pAA 1:15		4aAA 8:25	4pAA 1:00		5aAA 8:30
Studios Foyer				2aED 9:00			3aPPa 8:00 3aPPb 8:00	3pIDb 1:15		4aSCb 11:00	4pSC 1:00		5aSC 9:00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Studio 4
Studio 7
Service Area
Service Elevators
Elevators
Chartres Street
Studio 3
Studios Foyer
Studio 8
Service Corridor
Service Area
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Galerie 6
Studio 2
Studio 9
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Men
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Galerie 5
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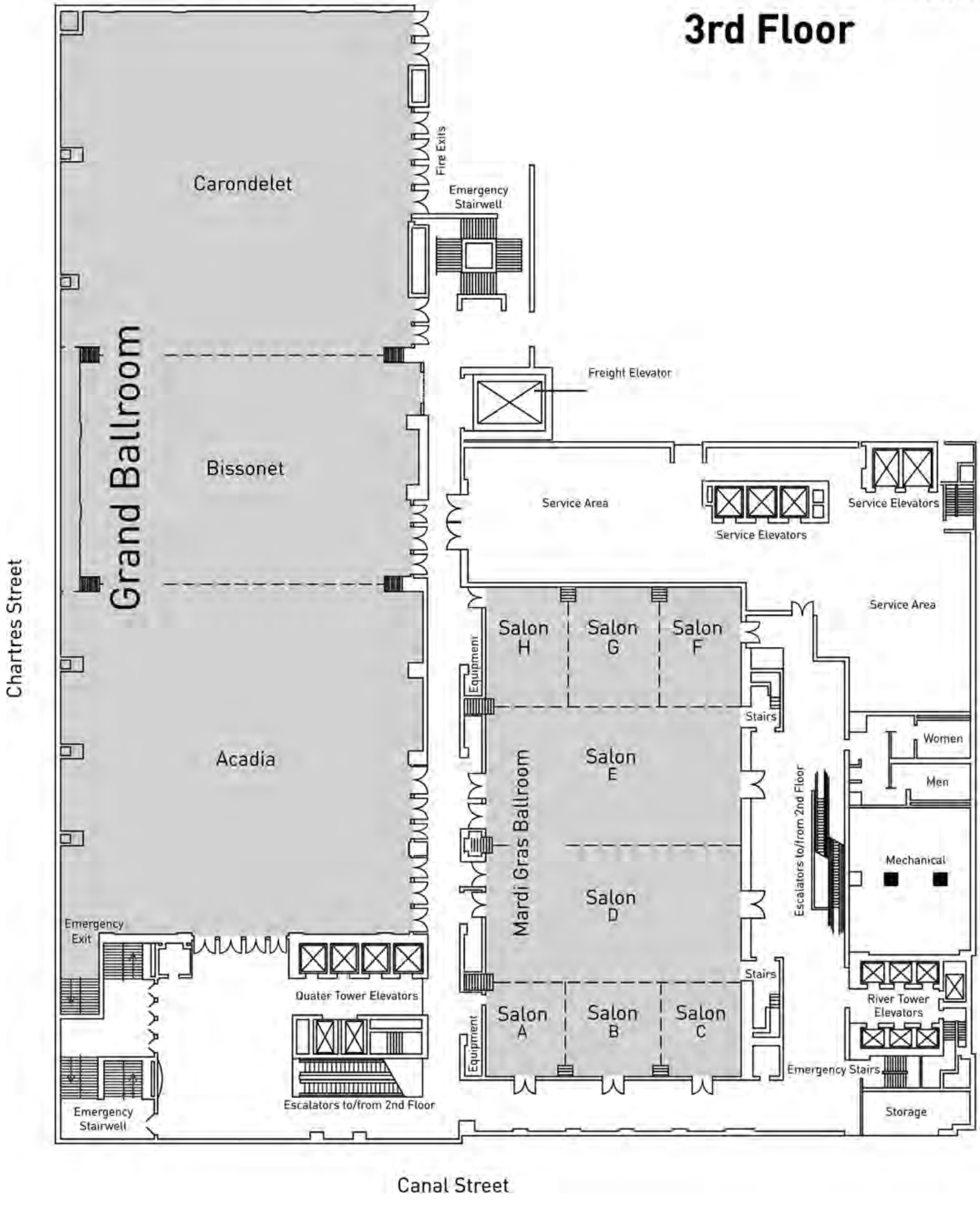
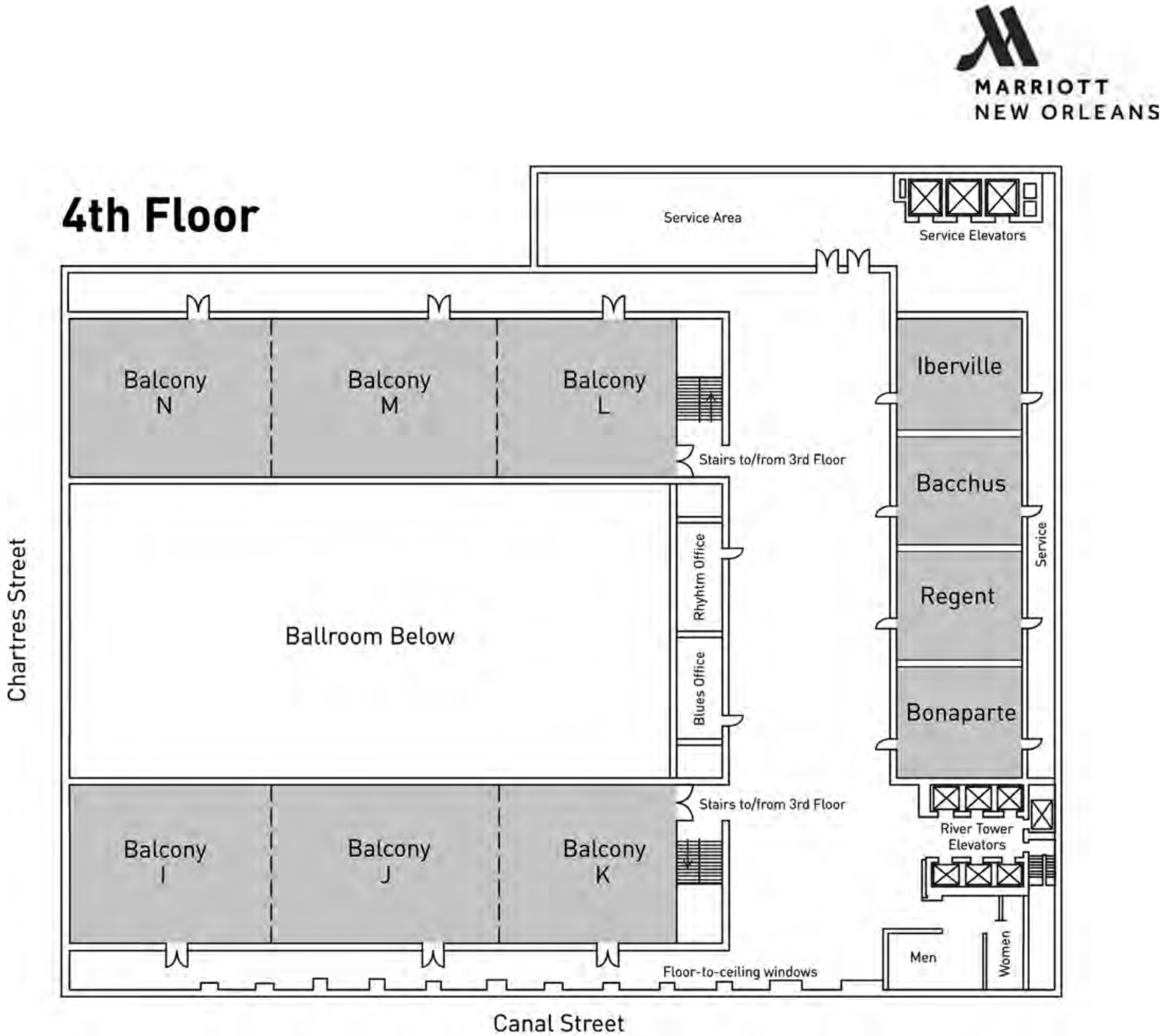
Men
Men
Booth
Women
Women
Emergency Stairwell
Quarter Tower Elevators
River Tower Elevators
Galerie 1
Galerie 4
Preservation Hall Foyer
Emergency Stairs
Escalators to/from Lobby, 3rd Floor
Emergency Stairwell
Canal Street

MARRIOTT
NEW ORLEANS

3rd Floor

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NEW ORLEANS
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Service Area
Service Elevators
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Balcony M
Balcony L
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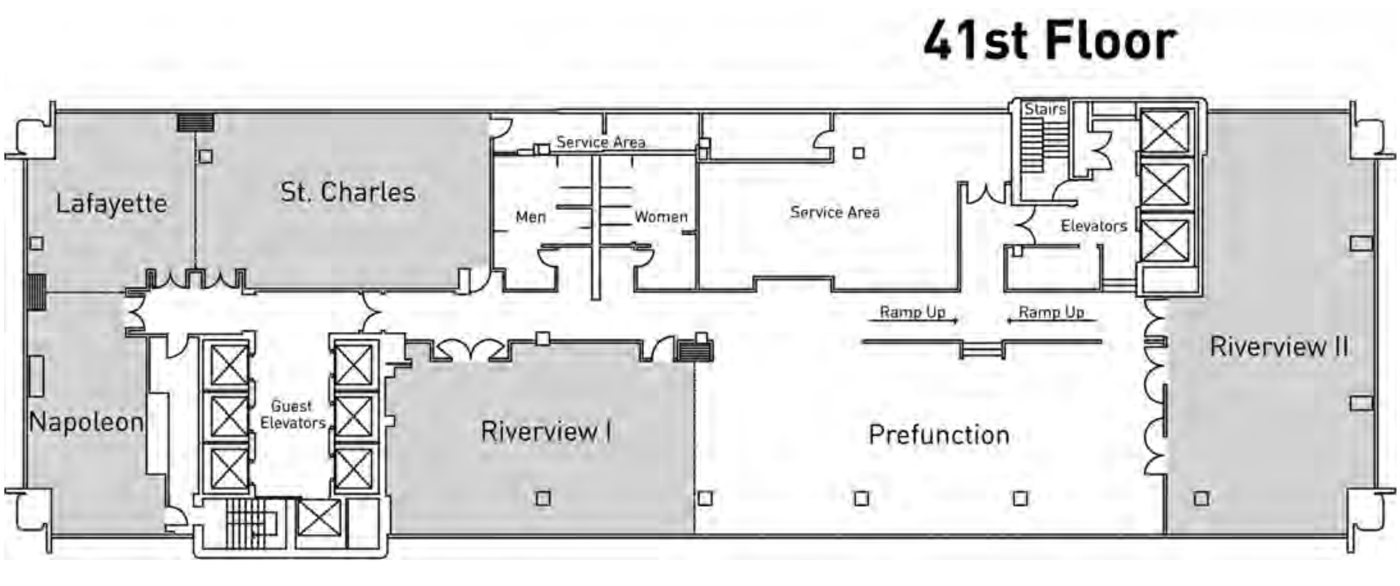
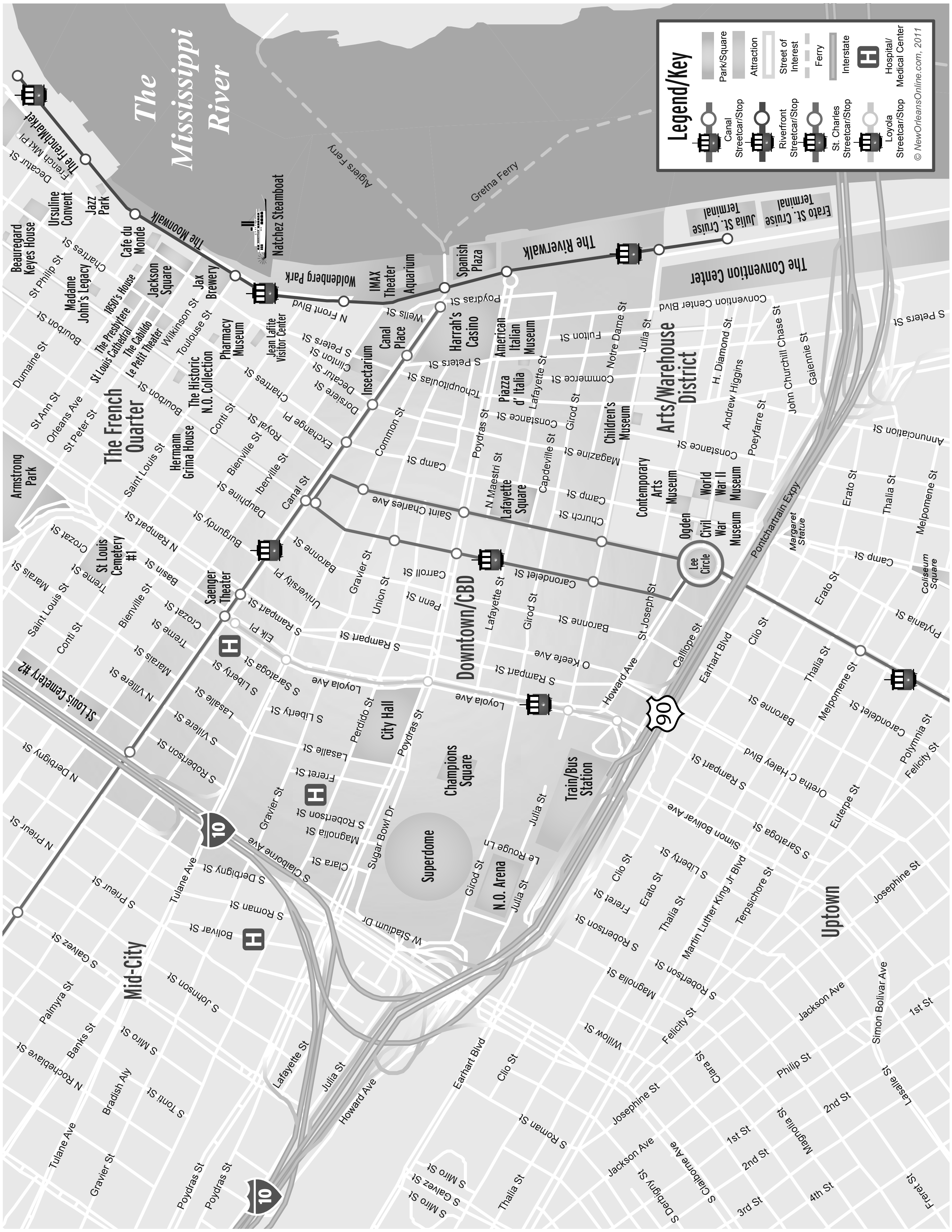
Bacchus
Service
Rhythm Office
Regent
Chartres Street
Ballroom Below
Blues Office
Bonaparte
Stairs to/from 3rd Floor
River Tower Elevators
Balcony I
Balcony J
Balcony K
Men
Women
Floor-to-ceiling windows
Canal Street


41st Floor
Stairs
Service Area
Lafayette
St. Charles
Men
Women
Service Area
Elevators
Ramp Up
Ramp Up
Riverview II
Guest Elevators
Napoleon
Riverview I
Prefunction



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The Cabildo
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Baronne St
Gravier St
Union St
Carroll St
Penn St
University Pl
S Rampart St
Elk Pl
S Saratoga St
S Liberty St
Loyola Ave
O Keefe Ave
St Joseph St
Howard Ave
Calliope St
Lasalle St
Perdido St
Freret St
S Robertson St
Magnolia St
Clara St
S Derbigny St
S Roman St
Bolivar St
S Claiborne Ave
Sugar Bowl Dr
Le Rouge Ln
W Stadium Dr
Tulane Ave
S Villere St
S Galvez St
S Johnson St
Palmyra St
Banks St
S Miro St
N Rocheblave St
Bradish Aly
S Tonti St
Lafayette St
Howard Ave
Earhart Blvd
Clio St
S Roman St
Thalia St
Jackson Ave
S Claiborne Ave
Josephine St
H. Diamond St.
Andrew Higgins
Poeyfarre St
John Churchill Chase St
Gaiennie St
S Peters St
Annunciation St
Erato St
Thalia St
Melpomene St
Pontchartrain Expy
Prytania St
Oretha C Haley Blvd
Simon Bolivar Ave
Martin Luther King Jr Blvd
Terpsichore St
Euterpe St
Polymnia St
Felicity St
Philip St
1st St
2nd St
3rd St
4th St
Jackson Ave
Simon Bolivar Ave
Willow St

TECHNICAL PROGRAM CALENDAR
174th Meeting of the Acoustical Society of America
4–8 December 2017

MONDAY MORNING

8:15 1aAA **Architectural Acoustics and Noise:** Impact of Entertainment Sound on Communities. Studio 9

8:00 1aAO **Acoustical Oceanography and Animal Bioacoustics:** Oceanographic Contributions to the Characteristics and Variability of the Underwater Soundscape. Salon F/G/H

9:00 1aEAa **Engineering Acoustics:** General Topics in Engineering Acoustics I. Balcony N

10:30 1aEAb **Engineering Acoustics:** General Topics in Engineering Acoustics II. Balcony N

7:50 1aNS **Noise and Physical Acoustics:** Supersonic Jet and Rocket Noise I. Studio 2

8:55 1aSA **Structural Acoustics and Vibration and ASA Committee on Standards:** Standards in Structural Acoustics and Vibration. Balcony L

9:00 1aSP **Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics:** Source Tracking with Microphone/Hydrophone Arrays I. Salon D

8:15 1aUW **Underwater Acoustics:** Underwater Acoustic Scattering and Reverberation. Salon E

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics and Psychological and Physiological Acoustics:** Perceived Diffuseness I. Studio 9

1:00 1pAB **Animal Bioacoustics and Acoustical Oceanography:** Bioacoustic Contributions to the Characteristics and Variability of Soundscapes, Underwater or Terrestrial. Salon F/G/H

1:00 1pAO **Acoustical Oceanography and Underwater Acoustics:** Acoustic Scattering from Hydrocarbons and Hydrothermal Vents. Balcony M

1:00 1pEAa **Engineering Acoustics:** General Topics in Engineering Acoustics III. Balcony N

2:45 1pEAb **Engineering Acoustics:** General Topics Engineering Acoustics IV. Balcony N

1:00 1pMU **Musical Acoustics:** Marching Band Instruments. Studio 4

1:00 1pNS **Noise and Physical Acoustics:** Supersonic Jet and Rocket Noise II. Studio 2

1:25 1pPA **Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics:** 30th Anniversary of the National Center for Physical Acoustics. Balcony I/J/K

3:30 1pSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration. Balcony L

2:00 1pSC **Speech Communication:** Voice, Tone, and Intonation (Poster Session). Acadia

1:00 1pSP **Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics:** Source Tracking with Microphone/Hydrophone Arrays II. Salon D

1:00 1pUW **Underwater Acoustics:** Underwater Acoustic Propagation: Models, Methods, and Statistics. Salon A/B/C

MONDAY EVENING

5:15 1eIDa **Interdisciplinary:** Special Presentation on The Clarinet in Early New Orleans Jazz. Salon E

7:00 1eIDb **Interdisciplinary:** Tutorial Lecture on Infrasound Phenomenology, Propagation, and Detection. Salon D

TUESDAY MORNING

8:00 2aAA **Architectural Acoustics and Musical Acoustics:** Performance Spaces for Modern Music. Studio 9

8:35 2aAB **Animal Bioacoustics, Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography:** In Memory of George Ioup: Acoustics in the Gulf of Mexico I. Salon E

8:25 2aBA **Biomedical Acoustics, Structural Acoustics, and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications I. Balcony M

9:00 2aEA **Engineering Acoustics:** Thermophone Transduction. Balcony N

9:00 2aED **Education in Acoustics:** Undergraduate Research Exposition (Poster Session). Studios Foyer

8:15 2aMU **Musical Acoustics:** Measurement Methods and Instrumentation for Musical Acoustics. Studio 4

9:00 2aPA **Physical Acoustics:** Sound Used as an Investigative Tool for Industrial Solutions. Balcony L

8:30 2aSA **Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in**

Acoustics, and Engineering Acoustics: Acoustic Metamaterials I. Salon A/B/C

9:00 2aSC **Speech Communication:** Articulatory and Acoustic Characteristics of Nasalization. Acadia

7:55 2aSP **Signal Processing in Acoustics and Underwater Acoustics:** Detection, Classification, Localization and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) I. Salon D

8:20 2aUW **Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics:** Sediment Characterization Using Direct and Inverse Techniques I. Salon F/G/H

TUESDAY AFTERNOON

1:15 2pAA **Architectural Acoustics and Psychological and Physiological Acoustics:** Perceived Diffuseness II. Studio 9

1:00 2pAB **Animal Bioacoustics, Signal Processing in Acoustics, and Acoustical Oceanography:** In Memory of George Ioup: Acoustics in the Gulf of Mexico II. Salon E

1:00 2pBA **Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications II. Balcony M

1:00 2pEA **Engineering Acoustics:** General Topics in Engineering Acoustics V. Balcony N

2:00 2pEDa **Education in Acoustics:** Acoustics Education Prize Lecture. Balcony I/J/K

3:15 2pEDb **Education in Acoustics:** General Topics in Acoustics Education. Balcony I/J/K

2:00 2pID **Interdisciplinary:** Guidance from the Experts: Applying for Grants and Fellowships. Studio 7

1:55 2pMUa **Musical Acoustics:** Cajun Music: Accordions, Culture, and History. Studio 4

5:00 2pMUb **Musical Acoustics:** Cajun Music Concert by the Savoy Family Band. Salon E

1:00 2pNSa **Noise, Architectural Acoustics, and ASA Committee on Standards:** Evaluation of Acoustics in Hospitals and Healthcare Facilities. Studio 2

3:30 2pNSb **Noise, Speech Communication, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Acoustics and its Role in Accessibility (e.g. Americans with Disabilities Act). Studio 2

1:00 2pPA **Physical Acoustics:** General Topics in Physical Acoustics I. Balcony L

1:00 2pSA **Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics and Engineering Acoustics:** Acoustic Metamaterials II. Salon A/B/C

1:00 2pSC **Speech Communication:** Speech Production (Poster Session). Acadia

1:00 2pSP **Signal Processing in Acoustics and Underwater Acoustics:** Detection, Classification, Localization and Tracking (DCLT) using Acoustics (and Perhaps Other Sensing Modalities) II. Salon D

1:00 2pUW **Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics:** Sediment Characterization Using Direct and Inverse Techniques II. Salon F/G/H

WEDNESDAY MORNING

7:45 3aAA **Architectural Acoustics:** Restaurant Acoustics. Studio 9

8:30 3aAB **Animal Bioacoustics:** General Bioacoustics. Salon F/G/H

11:00 3aAO **Acoustical Oceanography:** Munk Award Lecture. Salon E

8:25 3aBA **Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications III. Balcony M

8:30 3aED **Education in Acoustics:** Hands-On Acoustics Demonstrations for Middle- and High-School Students. Balcony I/J/K

7:40 3aID **Interdisciplinary and ASA Committee on Standards:** Practical Applications in Acoustics. Studio 7

8:45 3aMU **Musical Acoustics:** General Topics in Musical Acoustics. Studio 4

8:30 3aPA **Physical Acoustics and Biomedical Acoustics:** Acoustofluidics. Balcony L

8:00 3aPPa **Psychological and Physiological Acoustics:** Exploring the Perception of Sound (Poster Session). Studios Foyer

8:00 3aPPb **Psychological and Physiological Acoustics:** Perception and Physiology: Musicians, Musical Instruments, and the Body (Poster Session). Studios Foyer

8:30 3aSA **Structural Acoustics and Vibration:** Applications of Finite Element Analysis, Boundary Element Analysis, and Statistical Element Analysis Computational Methods. Balcony N

8:00 3aSC **Speech Communication and Education in Acoustics:** Teaching Phonetics and Speech

Science in the New Millennium: Challenges and Opportunities. Acadia

9:00 3aSP **Signal Processing in Acoustics and Underwater Acoustics:** Detection, Classification, Localization, and Tracking (DCLT) using Acoustics (and Perhaps Other Sensing Modalities) III. Salon D

7:40 3aUWa **Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics:** Sediment Characterization Using Direct and Inverse Techniques III. Salon E

8:00 3aUWb **Underwater Acoustics:** Session in Honor of Chester McKinney. Salon A/B/C

WEDNESDAY AFTERNOON

1:15 3pAA **Architectural Acoustics:** Speech Privacy Concerns in Open Plan Spaces. Studio 9

1:00 3pBA **Biomedical Acoustics:** Imaging I. Balcony M

1:00 3pIDa **Interdisciplinary:** Hot Topics: Hunt is Still Hot. Salon E

1:15 3pIDb **Interdisciplinary:** ASA Hunt Postdoctoral Research Fellows: Through the Years (Poster Session). Studios Foyer

1:15 3pNS **Noise and ASA Committee on Standards:** Urban Planning Using Soundscape I. Studio 2

1:00 3pPA **Physical Acoustics:** General Topics in Physical Acoustics II. Balcony L

1:00 3pSC **Speech Communication:** Clinical Populations (Poster Session). Acadia

1:00 3pSP **Signal Processing in Acoustics and Underwater Acoustics:** Detection, Classification, Localization, and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) IV. Salon D

WEDNESDAY EVENING

5:00 3eED **Education in Acoustics and Women in Acoustics:** Listen Up and Get Involved. Balcony I/J/K

THURSDAY MORNING

8:25 4aAA **Architectural Acoustics:** Speech Intelligibility in Reverberation and Noise. Studio 9

8:00 4aAB **Animal Bioacoustics:** General Biosonar. Salon F/G/H

8:00 4aAO **Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics:** Biological Effects on Seabed Geoacoustic Properties I. Salon A/B/C

8:25 4aBA **Biomedical Acoustics:** Ultrasound-Mediated Neuromodulation. Balcony M

7:55 4aNS **Noise and ASA Committee on Standards:** Urban Planning Using Soundscape II. Studio 2

9:00 4aPA **Physical Acoustics:** Infrasound, Atmospheric Sound Propagation, and Turbulence. Balcony L

9:00 4aPP **Psychological and Physiological Acoustics:** Psychoacoustics of Speech Perception in Noise, Localization, and Frequency Selectivity. Studio 4

8:00 4aSCa **Speech Communication:** The Southern States: Social Factors and Language Variation I. Studio 7

11:00 4aSCb **Speech Communication:** The Southern States: Social Factors and Language Variation II (Poster Session). Studios Foyer

8:00 4aSP **Signal Processing in Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Signal Processing in Acoustic Metamaterials. Salon D

8:00 4aUW **Underwater Acoustics:** Underwater Soundscapes and Noise: Measurement and Abatement. Salon E

THURSDAY AFTERNOON

1:00 4pAA **Architectural Acoustics:** Back to the Future: A Look at Multipurpose Spaces, How They've Changed & What's Next. Studio 9

1:00 4pAB **Animal Bioacoustics:** Neurophysiology of Echolocation. Salon F/G/H

1:00 4pAO **Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics:** Biological Effects on Seabed Geoacoustic Properties II. Salon A/B/C

1:00 4pBA **Biomedical Acoustics:** Imaging II. Balcony M

4:00 4pED **Education in Acoustics, Physical Acoustics, Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics:** Synthetic Aperture Sonar for Youngsters. Balcony I/J/K

1:00 4pNS **Noise, ASA Committee on Standards, and Structural Acoustics and Vibration:** Wind Turbine Noise. Studio 2

2:30 4pPA **Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics:** Acoustics of Detecting Gravitational Waves using LIGO. Balcony L

1:00 4pSC **Speech Communication:** Speech Perception and Word Recognition (Poster Session). Studios Foyer

1:00 4pSPa **Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics:** Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity I. Salon D

4:05 4pSPb **Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics:** Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity II. Salon D

1:00 4pUW **Underwater Acoustics:** Arctic Acoustics. Salon E

FRIDAY MORNING

8:30 5aAA **Architectural Acoustics:** Topics in Architectural Acoustics: Measurements, Modeling, and Isolation. Studio 9

9:00 5aAO **Acoustical Oceanography:** Topics in Acoustical Oceanography. Salon A/B/C

9:00 5aBA **Biomedical Acoustics:** Therapeutic Ultrasound. Balcony M

8:00 5aPA **Physical Acoustics and Structural Acoustics and Vibration:** Nonlinear Elasticity in Geomaterials. Balcony L

9:00 5aSC **Speech Communication:** Bilingual and Non-Native Speech Perception and Production (Poster Session). Studios Foyer

8:00 5aSP **Signal Processing in Acoustics:** Topics in Acoustic Signal Processing. Salon D

8:30 5aUW **Underwater Acoustics:** Underwater Measurements and Applications. Salon E

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 4 Dec, 7:30 a.m.	Executive Council	Studio 6
Mon, 4 Dec, 3:30 p.m.	Technical Council	Studio 6
Tue, 5 Dec, 7:00 a.m.	ASA Books	Studio 3
Tue, 5 Dec, 7:30 a.m.	Panel on Public Policy	Studio 10
Tue, 5 Dec, 11:45 a.m.	Editorial Board	Studio 6
Tue, 5 Dec, 12:00 noon	Student Council	Studio 10
Tue, 5 Dec, 12:30 p.m.	Prizes & Special Fellowships	Studio 3
Tue, 5 Dec, 1:30 p.m.	Meetings	Studio 8
Tue, 5 Dec, 4:00 p.m.	Newman Fund Advisory	Studio 3
Tue, 5 Dec, 5:00 p.m.	Women in Acoustics	Studio 6
Wed, 6 Dec, 6:45 a.m.	International Research & Education	Studio 2
Wed, 6 Dec, 7:00 a.m.	Archives & History	Studio 6
Wed, 6 Dec, 7:00 a.m.	College of Fellows	Studio 10
Wed, 6 Dec, 7:00 a.m.	Publication Policy	Studio 8
Wed, 6 Dec, 7:00 a.m.	Regional and Student Chapters	Riverview I
Wed, 6 Dec, 7:30 a.m.	Finance	Studio 3
Wed, 6 Dec, 11:00 a.m.	Medals and Awards	Studio 8
Wed, 6 Dec, 11:30 a.m.	Public Relations	Studio 3
Wed, 6 Dec, 12:00 noon	Membership	Studio 6
Wed, 6 Dec, 1:30 p.m.	AS Foundation Board	Studio 10
Wed, 6 Dec, 5:00 p.m.	Education in Acoustics	Studio 9
Wed, 6 Dec, 5:00 p.m.	TCAA Subcommittee on Speech Privacy	Studio 3
Thu, 7 Dec, 6:45 a.m.	International Liaison	Studio 3
Thu, 7 Dec, 7:30 a.m.	Tutorials, Short Courses, Hot Topics	Studio 8
Thu, 7 Dec, 7:30 a.m.	Investments	Studio 10
Thu, 7 Dec, 2:00 p.m.	Strategic Plan Champions	Studio 6
Thu, 7 Dec, 4:30 p.m.	Financial Affairs	Studio 6
Thu, 7 Dec, 4:30 p.m.	Member Engagement and Diversity	Studio 8
Thu, 7 Dec, 4:30 p.m.	Outreach	Studio 10
Thu, 7 Dec, 4:30 p.m.	Publications and Standards	Studio 3
Fri, 8 Dec, 7:00 a.m.	Technical Council	Studio 6
Fri, 8 Dec, 11:00 a.m.	Executive Council	Studio 6

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 5 Dec, 4:30 p.m.	Engineering Acoustics	Studio 7
Tue, 5 Dec, 7:30 p.m.	Acoustical Oceanography	Salon A/B/C
Tue, 5 Dec, 7:30 p.m.	Animal Bioacoustics	Salon F/G/H
Tue, 5 Dec, 7:30 p.m.	Architectural Acoustics	Studio 9
Tue, 7 Dec, 7:30 p.m.	Musical Acoustics	Studio 4
Tue, 5 Dec, 7:30 p.m.	Physical Acoustics	Balcony L
Tue, 5 Dec, 7:30 p.m.	Psychological and Physiological Acoustics	Balcony M
Tue, 5 Dec, 7:30 p.m.	Structural Acoustics and Vibration	Studio 7
Wed, 6 Dec, 7:30 p.m.	Biomedical Acoustics	Balcony M
Wed, 6 Dec, 7:30 p.m.	Signal Processing in Acoustics	Salon D
Thu, 7 Dec, 7:30 p.m.	Noise	Studio 2
Thu, 7 Dec, 7:30 p.m.	Speech Communication	Salon A/B/C
Thu, 7 Dec, 7:30 p.m.	Underwater Acoustics	Salon F/G/H

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 4 Dec, 7:00 p.m.	ASACOS Steering	Studio 3
Tue, 5 Dec, 7:30 a.m.	ASACOS	Studio 8
Tue, 5 Dec, 9:00 a.m.	S3/SC1 WG6	Studio 6

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 4-7 Dec 7:30 a.m. - 5:00 p.m. Fri, 8 Dec 7:30 a.m. - 12:00 noon	Registration	Acadia Foyer
Mon-Thu, 4-7 Dec, 7:00 a.m. - 5:00 p.m. Fri, 8 Dec, 7:00 a.m. - 12:00 noon	E-mail	Preserva-tion Foyer
Mon-Thu, 4-7 Dec, 7:00 a.m. - 5:00 p.m. Fri, 8 Dec, 7:00 a.m. - 12:00 noon	Internet Zone	Preserva-tion Foyer
Mon-Thu, 4-7 Dec, 7:00 a.m. - 5:00 p.m. Fri, 8 Dec 7:00 a.m. - 12:00 noon	A/V Preview	Studio 5
Mon-Fri, 4-8 Dec, 10:00 a.m. - 11:00 a.m.	A.M. Coffee Break	Preserva-tion Foyer
Mon-Thu, 4-8 Dec 8:00 a.m. - 10:00 a.m.	Accompanying Persons	St. Charles
Sun, 3 Dec, 1:00 p.m. - 5:00 p.m. Mon, 4 Dec 8:30 a.m. - 12:30 p.m.	Short Course	Bonaparte
Mon, 4 Dec 1:00 p.m. to 3:30 p.m.	Technical Tour	Meet at Marriott Lobby
Mon, 4 Dec 5:00 p.m. - 5:30 p.m.	New Student Orientation	Salon F-H
Mon, 4 Dec 5:30 p.m. - 7:00 p.m.	Student Meet and Greet	Riverview I
Tue, 5 Dec 6:00 p.m. - 7:30 p.m.	Social Hour	Carondelet
Wed, 6 Dec, 11:45 a.m. - 1:45 p.m.	Women in Acoustics Luncheon	Riverview I
Wed, 6 Dec, 3:30 p.m.	Annual Membership Meeting	Bissonet
Wed, 6 Dec, 3:30 p.m. - 5:00 p.m.	Plenary Session/Awards Ceremony	Bissonet
5:30 p.m. - 6:30 p.m.	Hunt Fellowship/Early Career Leadership Campaign Reception	Riverview Prefunction
Wed, 6 Dec, 6:30 p.m. - 8:30 p.m.	Hunt Fellowship Celebration / Early Career Leadership campaign Dinner	Riverview II
Wed, 6 Dec, 1:30 p.m. - 3:00 p.m.	Early Career Publishing Workshop	Salon F/G/H
Wed, 6 Dec, 6:00 p.m. - 8:00 p.m.	Student Reception	Riverview I
Wed, 6 Dec, 8:00 p.m. - 12:00 midnight	ASA Jam	One Eyed Jacks
Thu, 7 Dec, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Riverview I
Thu, 7 Dec, 6:00 p.m. - 7:30 p.m.	Social Hour	Carondelet

174th Meeting of the Acoustical Society of America

The 174th meeting of the Acoustical Society of America will be held Monday through Friday, 4-8 December 2017 at the New Orleans Marriott Hotel, New Orleans, Louisiana.

SECTION HEADINGS

1. HOTEL INFORMATION

The New Orleans Marriott 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 New Orleans Marriott Hotel, 555 Canal Street, New Orleans, Louisiana 70130-2349 for information about room availability.

2. TRANSPORTATION AND TRAVEL

The Louis Armstrong New Orleans International Airport, (Airport Code MSY) is served by 15 international and domestic airlines. For further information visit www.flymsy.com. The airport is approximately 16 miles from the New Orleans Marriott, at the boundary of the French Quarter and the Central Business District.

Taxi: Taxicab booths are located on the first level of the Terminal outside of Baggage Claim Belts 1 and 14. Passengers must wait in line at one of these booths for taxi service. Taxi rides cost $36.00 from the airport to the Central Business District (CBD) and French Quarter for up to two passengers. For three or more passengers, the fare will be $15.00 per passenger. Taxis are required to accept credit card payments.

Airport Shuttle: Shuttle service is available from the airport to hotels and various other locations in the New Orleans for $24.00 (per person, one-way) or $44.00 (per person, round-trip). These fares include three (3) bags per person. Additional baggage may be subject to additional fees. Passengers can purchase tickets at the airport at Airport Shuttle ticket booths located on the first level throughout the Baggage Claim area. Those needing Wheelchair-Accessible Service should also call 1-866-596-2699 for assistance. Visit www.airportshuttleneworleans.com for more information.

Car Rental: Nine car rental agencies serve the airport. To get to the Consolidated Rental Car Facility, walk to the West Terminal Baggage Claim (Claims 12-14) and proceed outside the building. To the right is a covered walkway leading to the customer service building. Passengers with special needs should contact MVI Field Services at (615) 318-3108.

New Orleans is at the intersection of Interstate 55 from the north and Interstate 10 from the east and west. It can be reached by car in 8 hours or less from Atlanta, GA, Jacksonville, FL, Dallas, TX, Austin, TX, and Nashville, TN.

Hotel parking is $47.56/day, $19 for up to 6 hours, and $24 for 6 to 10 hours.

3. MESSAGES FOR ATTENDEES

A message board will be located in the Acadia 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.

4. 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, 4 December, at 7:30 a.m. in the Acadia Foyer (see floor plan on page A11).

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 $625 for members of the Acoustical Society of America; $750 for non-members, $175 for Emeritus members (Emeritus status pre-approved by ASA), $365 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $150 for ASA Student members, $250 for students who are not members of ASA, and $175 for accompanying persons.

One-day registration is available at $365 for members and $425 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 $750 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 2018 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the Member full-week or one-day registration fees. Nonmember invited speakers who participate in the meeting only on the day of their presentation may register for $100. The registration fee for nonmember invited speakers who wish to participate for more than one day is $215 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.

5. ACCESSIBILITY

If you have special accessibility requirements, please inform ASA (1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@acousticalsociety.org) at a minimum of thirty days in advance of the meeting. Please provide a cell phone number, email address, and detailed information including the nature of the special accessibility so that we may contact you directly.

6. TECHNICAL SESSIONS

The technical program includes 96 sessions with 922 abstracts scheduled for presentation during the meeting.

A floor plan of the Marriott Hotel appears on page A11. 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.

7. TECHNICAL SESSION DESIGNATIONS

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

1-Monday, 4 December
2-Tuesday, 5 December
3-Wednesday, 6 December
4-Thursday, 7 December
5-Friday, 8 December

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.

8. HOT TOPICS SESSION

The Hot Topics session (3pIDa) will be held on Wednesday, 6 December, at 1:00 p.m. in Salon E. Papers will be presented on current topics in the fields of Architectural Acoustics, Noise, and Underwater Acoustics.

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

The 2017 Rossing Prize in Acoustics Education will be presented to Robert Celmer, University of Hartford, at the Plenary Session on Wednesday, 6 December. Dr. Celmer will present the Acoustics Education Prize Lecture titled "Student-centered acoustical engineering education at the University of Hartford" on Tuesday, 5 December, at 2:00 p.m. in Session 2pEDa in Balcony I/J/K.

10. WALTER MUNK AWARD AND MUNK AWARD LECTURE

Andone Lavery of the Woods Hole Oceanographic Institution has been named recipient of the 2017 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.

Andone Lavery will present the Munk Award Lecture on Wednesday, 6 December, at 11:00 a.m. in Session 3aAO in Salon E. The award will be presented to Dr. Lavery following the lecture and she will be introduced during the Plenary Session on Wednesday afternoon.

11. TUTORIAL LECTURE ON SONIC BOOMS

A tutorial presentation titled "Infrasound Phenomenology, Propagation, and Detection" will be given by Roger M. Waxler of the University of Mississippi on Monday, 4 December, at 7:00 p.m. in Salon D.

This is tutorial will present an overview of the generation of infrasound by both natural and anthropomorphic sources and of the subsequent propagation and detection of infrasonic signals Lecture notes will be available at the meeting 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).

12. SHORT COURSE ON ULTRASOUND CONTRAST AGENTS

A short course on Ultrasound Contrast Agents will be given by Tyrone M. Porter, Boston Univerisity, in two parts: Sunday, 3 December, from 1:00 p.m. to 5:00 p.m. and Monday, 4 December, from 8:30 a.m. to 12:30 p.m. in the Bonaparte Room.

This course will cover the various formulations of Ultrasound Contrast Agents, experimental and theoretical approaches for studying their response to acoustic excitation, and the variety of biomedical applications that benefited from their development. Instructional materials and coffee breaks are included. Onsite registration at the meeting will be on a space-available basis.

13. EARLY CAREER PUBLISHING WORKSHOP

Publication of one's technical results is important to many people for many reasons, but it is especially important to early career academics. In this workshop, the JASA Editor-in-Chief and a number of Associate Editors will provide such a perspective by having the participants handle a mock submission from the viewpoint of an author, an Associate Editor, and a reviewer.

The workshop will be held on Wednesday, 6 December 1:30 p.m. to 3:00 p.m., in Salon F/G/H.

14. UNDERGRADUATE RESEARCH POSTER EXPOSITION

The Undergraduate Research Exposition will be held Tuesday, 5 December, 9:00 a.m. to 12:00 noon in session 2aED in the Studios Foyer.

The 2017 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. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at the Marriott. The schedule and rooms for each Committee meeting are given on page A10.

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.

16. TECHNICAL TOUR

Monday, 4 December, 1:00 p.m. to 3:30 p.m. - Tour Fee: USD $20

A walking tour of recently renovated performing arts spaces of the New Orleans Theater District will be held on Monday, 4 December, from 1:00 p.m. to 3:30 p.m. The tour will explore the three grand theaters of New Orleans, the Orpheum, Saenger and Joy, examining their histories and restorations.

The group will leave from the Marriott Hotel lobby at 12:45 p.m., and walk five blocks along Canal St. to the theater district. In case of inclement weather, transportation will be provided. The cost is USD $20 and is limited to 30 participants.

17. ANNUAL MEMBERSHIP MEETING

The Annual Membership Meeting of the Acoustical Society of America will be held at 3:30 p.m. on Wednesday, 4 December 2017, in the Bissonet Room at the New Orleans Marriott Hotel, 555 Canal Street, New Orleans, LA 70130.

18. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 6 December, at 3:30 p.m. in the Bissonet Room.

ASA scholarship recipients and the 2017 Munk Award recipient will be introduced. The Rossing Prize in Acoustics Education will be presented to Robert D. Celmer. The Pioneers of Underwater Acoustics medal will be presented to Michael J. Buckingham, the Silver Medal in Physical Acoustics will be presented to Evegenia Zabolotskaya, and the Wallace Clement Sabine Medal will be presented to David Griesinger. Certificates will be presented to Fellows elected at the Boston meeting. See page 2646 for a list of fellows.

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

19. HUNT FELLOWSHIP RECOGNITION AND CAMPAIGN FOR ASA EARLY CAREER LEADERSHIP

The Hunt Fellowship has been an outstanding feature of the Acoustical Society of America for 40 years. Recipients of the Fellowship have made major contributions to acoustical science and service to the Society. An event to commemorate and celebrate this anniversary will be held at the New Orleans meeting.

In conjunction with this celebration, the Acoustical Society Foundation Fund, with support of the Executive Council, is initiating a fund raising campaign. Early Career Leadership (CACEL) Campaign for ASA for development of talent within the Society.

A celebratory reception (with cash bar) open to all is scheduled for Wednesday 6 December 2017, starting at 5:30 p.m., followed by a special dinner at 6:30 p.m., in conjunction with the celebration of the Hunt Postdoctoral Research Fellowship's 40th Anniversary. Please check at the ASA Registration Desk for information about the availability of tickets for the dinner.

20. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will not be held at the New Orleans 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 A21 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 Manage r for further information about these groups, or about the ASA Standards Program in general, at the following address: Neil Stremmel, 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

21. COFFEE BREAKS

Morning coffee breaks will be held each day from 10:00 a.m. to 11:00 a.m. near the Preservation Hall Foyer on the third floor.

22. A/V PREVIEW ROOM

Studio 5 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.

23. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The New Orleans 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://asa.scitation/journal/poma

24. 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 on the third floor near the Acadia Foyer.

Tables with power cords will be set up near the Preservation Hall Foyer on the third floor for attendees to gather and to power-up their electronic devices.

25. SOCIALS

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. in the Carondelet/Bissonet Room.

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

26. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 7 December, at 12:00 noon in Riverview I on the 41st floor. The speaker will be Gabriela González, Professor of Physics and Astronomy at Louisiana State University.

Dr. González is a founding member of the Laser Interferometer Gravitational-Wave Observatory (LIGO) Scientific Collaboration and serves as the spokesperson for the group. The title of her talk is "Listening to the Universe with Gravitational Waves." In 2015, one hundred years after Einstein's unveiling of general relativity, the LIGO detectors picked up a faint chirp signal, the result of the merging of two black holes over one billion years ago. This marked the first direct detection of the ripples of space-time predicted by Einstein's theory. Dr. Gonzalez's research deals with the characterization of noise in the supremely sensitive LIGO system. She is a fellow of three professional societies, and the recipient of numerous awards in Physics and Astronomy.

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 6 December. The cost is USD $30.00 per ticket.

27. 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 on Monday, 4 December, from 5:00 p.m. to 5:30 p.m. in Salons F/G/H for all students to learn about the activities and opportunities available for students at the New Orleans meeting. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in Riverview I on the 41st floor where refreshments and a cash bar will be available.

A Fellowship and Grant Panel titled Guidance from the Experts: Applying for Grants and Fellowships, will be held in session 2pID on Tuesday, 5 December, from 2:00 p.m. to 3:30 p.m. in Studio 7. 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, 6 December, from 6:00 p.m. to 8:00 p.m. in Riverview I on the 41st floor. 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.

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.

28. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 6 December, in Riverview I on the 41st floor. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 5 December. The fee is USD $30 for non-students and USD $15 for students.

29. JAM SESSION

You are invited to One Eyed Jacks on Wednesday night, 6 December, 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.

30. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the New Orleans meeting. The on-site registration fee for accompanying persons is USD $175. A hospitality room for accompanying persons will be open in the St. Charles room on the 41st floor at the Marriott from 8:00 a.m. to 10:00 a.m. Monday through Friday. Registration entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

The unique blend of traditions in New Orleans has yielded a city rich in music, architecture and cultural institutions. Within a short walk of the hotel are dozens of museums and landmarks as well as a broad range of culinary experiences. There are also a variety of excellent tours to choose from. The New Orleans Convention and Visitors Bureau (www.neworleanscvb.com) is a great resource. Check out the link www.neworleanscvb.com/press-media/story-ideas/ for an overview on a range of topics.

31. WEATHER

New Orleans has a subtropical climate with pleasant year-round temperatures. Average high and low temperatures in December are 64 F (18 C) and 50 F (9C), respectively. Rainfall is common in New Orleans and occurs on an average of ten days in December. Carrying a small foldable umbrella may be useful for occasional showers.

32. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Natalia Sidorovskaia, Chair; David P. Knobles, Acoustical Oceanography; Aaron Thode, Animal Bioacoustics; Ian Hoffman, Shane Kanter, Architectural Acoustics; Siddhartha Sikdar and Kang Kim, Biomedical Acoustics; Peggy B. Nelson, Andrew Oxenham, Daniel Russell, Education in Acoustics; Kenneth M. Walsh, Engineering Acoustics; Whitney Coyle, Peter Rucz, Musical Acoustics; Michael Haberman, Kevin Lee, Physical Acoustics; Anna Diedesch, Christopher Brown, Psychological and Physiological Acoustics; Kai Gemba, Zoi-Heleni Michalopoulou, Signal Processing in Acoustics; Melissa Baese-Berk, Kristin Van Engen, Speech Communication; Benjamin Shafer, Structural Acoustics and Vibration; Jason Sagers, Underwater Acoustics; Kelly Whiteford, Student Council.

33. MEETING ORGANIZING COMMITTEE

Joel Mobley, Chair; Natalia Sidorovskaia, Technical Program Chair; Josette Fabre, Juliette Ioup, Student Coordinators; Cecille Labuda, Likun Zhang, Signs; David Woolworth, Andi Petculescu, Technical Tours.

34. PHOTOGRAPHING AND RECORDING

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

35. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2017 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*.

36. 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.

37. 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.

38. 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.

39. 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

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, San Diego, CA, 30 November–4 December 2019
179th Meeting, Chicago, Illinois, 11–15 May 2020
180th Meeting, Cancun, Mexico, fall 2020

ASA Exhibit

At the 175th Meeting of the Acoustical Society of America

7-11 May 2018
Hyatt Regency Minneapolis, Minnesota, USA

The 175th Meeting of the Acoustical Society of America will bring together experts from the all fields of acoustics and will provide a forum for the open exchange of scientific information.

The 175th Meeting will consist of plenary lectures, invited and contributed papers, poster presentations and exhibits.

The ASA Meeting will be highlighted by an exhibit featuring displays with instruments, materials, and services for the acoustics and vibration community. The exhibit will be conveniently located near the registration area and meeting rooms and will open with a reception on Monday evening, 7 May, and will close Wednesday, 9 May, at noon.

Morning and afternoon refreshments will be available in the exhibit area.

For ASA Exhibit information:	For information on the ASA Meeting:
Bob Finnegan, ASA Exhibits Manager AIP Publishing LLC 1305 Walt Whitman Rd, Suite 300 Melville, NY 11747 Tel: 516-576-2433 Fax: 631-923-2872 Email: rfinnegan@aip.org	Acoustical Society of America 1305 Walt Whitman Rd, Suite 300 Melville, NY 11747 Tel: 516-576-2360 Fax: 631-923-2875 Email: asa@acousticalsociety.org Web site: http://AcousticalSociety.org/

Topics to be covered include:

- Acoustical Oceanography
- Animal Bioacoustics
- Architectural Acoustics
- Biomedical Acoustics
- Engineering Acoustics
- Musical Acoustics
- Noise
- Physical Acoustics
- Physiological Acoustics
- Psychological Acoustics
- Signal Processing in Acoustics
- Speech Communication
- Structural Acoustics and Vibration
- Underwater Acoustics

Session 1aAA

Architectural Acoustics and Noise: Impact of Entertainment Sound on Communities

David Woolworth, Cochair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

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

Chair's Introduction—8:15

Invited Papers

8:20

1aAA1. Challenges of entertainment sound in communities. Gary W. Siebein, Gary Siebein, Hyun Paek, Marylin Roa, Jennifer R. Miller, and Keely Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

There are three basic challenges of entertainment sound in communities. First is how to measure the sound. People often clearly hear individual notes of music being played, individual beats from drums, and other lower frequency percussive instruments and sounds of words sung by performers at locations away from the facility. Equivalent sound levels taken over varying periods of time, LA Fast, LA slow, and other experimental metrics will be compared with the audibility of the sounds by people at remote listening locations. Second, it is often necessary to correct the measurements for the effects of background noise when equivalent sound levels are measured because the short term transient sounds that contain the musical information have been "averaged away" and the resulting sound levels are in the vicinity of the background sounds even though the words and music are plainly audible. Third is that methods to effectively contain the sounds from outdoor amplified entertainment facilities including operational controls, infrastructure controls, and administrative controls should be considered. All three challenges must be addressed to optimize the compatibility of the facility with the community.

8:40

1aAA2. Noise impact modeling in two residential areas from amplified music. David Manley, Benjamin Bridgewater, Ted Pitney, Ian Patrick, and Edward Logsdon (D. L. Adams Assoc., Inc, 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

D.L. Adams Associates has recently worked on two outdoor music venue projects, both in close proximity to residences in which venue owners were concerned about noise impact to the neighboring community. The first project involved the renovation of a city owned garden with the addition of a large stage and audience area. Residences are located within 450 feet of the proposed stage. Modeling was completed to evaluate noise impact, and mitigation recommendations were provided. The second project involved a privately owned sports facility which was leased to provide an Electronic Dance Music (EDM) festival over the course of a weekend. After reading about complaints of many such EDM festivals, the facility Owner wanted to understand the potential noise impact to the neighbors prior to the event. Environmental noise modeling of the facility and anticipated musical acts was completed to predict sound levels in the surrounding neighborhoods and demonstrate to the facility Owner the potential for complaints from the festival.

9:00

1aAA3. Interpreting conflicting noise ordinances imposed upon Merriweather Post Pavilion. Josh Curley and Scott Harvey (Phoenix Noise & Vib., 5216 Chairmans Court, Ste. 107, Frederick, MD 21703, sharvey@phoenixnv.com)

Merriweather Post Pavilion in Columbia, Maryland, has been host to numerous concerts and music festivals each year since it opened in 1967. Over the years, noise complaints regarding Merriweather have not been uncommon during musical acts. In 2013, a noise ordinance was placed on the venue that allows concert events to generate a noise level up to 95 dBA at a $^1/_4$-mile radius from the main stage between the hours of 9:00 AM and 11:00 PM, and 72.5 dBA at residential property lines. Between 11:00 PM and 11:30 PM, main stage noise levels must not exceed 55 dBA as measured both within the $^1/_4$-mile radius and at residential property lines. Currently, the residence closest to the main stage is approximately 1,400 feet away, just outside of the $^1/_4$-mile 95 dBA allowance; however, residential construction is planned in the next few years as close as 750 feet behind the main stage. Once these residential buildings are occupied, how will the noise ordinance be applied if main stage noise is greater than the 72.5 dBA allowance at residential buildings yet below the 95 dBA limit within the $^1/_4$-mile radius? And, how will the likely complaints regarding audible low frequency noise inside the residences, which are not addressed by the noise ordinance, be handled?

9:20

1aAA4. Consistency and accountability: First steps in a quest for compromise between venue and community. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

This case study examines the first steps in reducing several years of tension between an active outdoor venue, with more than 60 performances each summer, and the surrounding community in a small coastal city in New Hampshire. Continuous sound level monitoring and realtime visual feedback provide means of achieving consistent (though somewhat arbitrary) sound levels throughout and across performances, thus eliminating a variable in the evaluation of impacts on the community. Sound level data, along with feedback from the community, will be used to develop mitigation strategies for future seasons. Details of the monitoring and feedback systems and lessons learned from the 2017 season will be shared.

Contributed Paper

9:40

1aAA5. Noise assessment for residential developments near places of entertainment in San Francisco. Jordan L. Roberts (none, Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, jordan.roberts@cmsalter.com)

When new multi-family residential developments are proposed in San Francisco, the City's Entertainment Commission identifies whether there are places of entertainment (POE) within 300 feet. A POE includes businesses such as music venues, clubs, and bars. The Commission's goal is to reduce complaints from new nearby residents about noise generated by existing entertainment venues. If there are too many complaints, a POE might be forced to close its business, which would have a negative impact on the City's various entertainment districts. This assessment addresses noise that is distinct from the noise associated with a typical environmental noise study (which addresses noise from street traffic, rail activity, etc.). Rather than put the onus on existing businesses, the housing developer is asked to consider bass-heavy nighttime noise sources that future residents could perceive. Thus, some facades of proposed buildings might need to provide greater noise reduction than typical construction, often times through increased window STC rating recommendations. The measurements, calculations, and mitigations of these assessments are to be presented through select data from multiple case studies.

Invited Papers

9:55

1aAA6. Typical outdoor audio system designs. Andrew N. Miller (BAi, LLC, 4006 Speedway, Austin, TX 78758, amiller@baiaustin.com)

Trends in audio system designs for outdoor entertainment venues impact the sound in surrounding communities. The author explores typical audio system designs for outdoor amplified music venues and high school football stadiums in and around Austin, Texas. The consequences of particular system designs will be discussed as they relate to sound in surrounding communities. Given special consideration are characteristics of sound at the property line of venues employing loudspeaker array and point-source systems.

10:15–10:30 Break

10:30

1aAA7. Impact and control practices of bar and pub sound in densely populated cities. Andy Chung (Smart City Maker, Smart City Maker, Copenhagen, Denmark, ac@smartcitymaker.com), W. M. To (Macao Polytechnic Inst., Macao, Macao), K. K. Iu (Supreme Acoust. Res. Ltd., Hong Kong, China), and Maurice Yeung (Macau Instituto de Acústica, Macau, China)

While patrons are enjoying the happy moments in bars and pubs, the sound generated from these premises may not be wanted by the neighboring community, in particular, during the sensitive hours. The situation becomes more challenging in a densely populated city, where various constraints exist in attempting to alleviate the noise impacts from propagation paths and the receiving ends. Controlling at source is considered to be the most effective approach. This paper presents the prevailing practices in Hong Kong, in China's Bay Area, to control the noise impacts associated with the bars and pubs, and illustrates with examples.

10:50

1aAA8. Benefits of managing source sound levels in amplified indoor entertainment venues. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper investigates sound levels on stage and in the audience from field measurements and the literature for small and midsize amplified venues for various entertainment types. The effect of sound levels are also briefly examined in regard to hearing and comfort of patrons and employees. Furthermore, reduction in sound level internally is examined as a community noise control measure relative to soundproofing.

1a MON. AM

11:10

1aAA9. Entertainment sounds, the soundscape, and resources. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Assessing the impact of entertainment sounds in communities will need a multifaceted analysis. An evaluation process has to be considered that will cover the different facets with regard to people's understanding about an acoustic environment where the impact of entertainment sound will occur. The "meeting of music and environment" discussed by R. M. Schafer opens the mind in many directions to learn about the different aspects and related impacts in such a soundscape that is relying on given activities in an area that will lead to a harmonized or an acoustically unbalanced situation. Understanding human auditory scene analysis and the important role of auditory attention forces to outline soundscape assessment methods and to come to enhanced methodologies for inventing action parameters with regard to a sounding community. This paper will introduce regulations to harmonize the different interests regarding interventions through entertainment sounds in communities but also introduce the soundscape approach to offer new features with regard to a balanced and harmonic soundscape addressing entertainment sounds as a resource.

11:30

1aAA10. Case study of an outdoor music festival in a rural community. Robert M. Lilkendey (RML Acoust., LLC, 14688 NW 150th Ln., Alachua, FL 32615, rob@rmlacoustics.com)

A case study will be presented of the author's involvement in assisting Okeechobee County, Florida, with the development of acoustical criteria and monitoring of live music events associated with a four-day outdoor music festival that takes place annually in a rural part of the county. The presentation will include the results of these efforts, as well as a discussion of the collaborative process that continues to take place between the promoter, promoter's acoustical consultant, the county planning and zoning department, residents, elected county officials, law enforcement, and the author to work toward a solution for future festivals that balances the monetary benefits for the entire county with the needs of the relatively small percentage of county residents adversely affected by the noise.

MONDAY MORNING, 4 DECEMBER 2017 SALON F/G/H, 8:00 A.M. TO 10:05 A.M.

Session 1aAO

Acoustical Oceanography and Animal Bioacoustics: Oceanographic Contributions to the Characteristics and Variability of the Underwater Soundscape

David R. Barclay, Cochair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada

Bruce Martin, Cochair
JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada

Chair's Introduction—8:00

Invited Paper

8:05

1aAO1. Tidal influence on underwater soundscape characteristics in a shallow-water environment off Taiwan. Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Chi-Fang Chen, Chih-Hao Wu, Dai-Hua Liu (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Passive acoustic monitoring has been conducted in shallow-water environment off the coast of west Taiwan using bottom-mounted hydrophones to obtain baseline soundscape information prior to large scale wind farm constructions. Analyses of long-term acoustic datasets were conducted in three distinctive one-octave bands: 150-300 Hz, 1,200-2,400 Hz, and 3,000-6,000 Hz. Initial study was able to link the 150-300 Hz band to passages of cargo vessels. However, further research using time-frequency analysis showed an approximate 6-hour cycle of strong occurrence of this noise band, which corresponds particularly well with the local semidiurnal tidal cycle. The results suggest that at least part of the high noise levels between the high- and low-water periods in the 150-300 Hz octave band was due to flow noise from tidal movements. The results underlines the importance to consider oceanographic processes such as tidal and wave movements when studying the contribution of specific acoustic sources to the underwater soundscape.

Contributed Papers

8:20

1aAO2. Long-term measurements of the ice-free underwater noise field directionality in the shallow Beaufort Sea. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), Alexander Conrad, and Katherine H. Kim (Greeneridge Sci., Inc., Santa Barbara, CA)

Between 2007 and 2014, over 35 Directional Autonomous Seafloor Acoustic Recorders (DASARs) were deployed over a 280 km swath of the Beaufort Sea continental shelf (20-55 m depth) during the open-water season, in order to monitor the westward bowhead whale migration. DASARs have one omnidirectional pressure sensor and two orthogonal particle velocity sensors that permit instantaneous measurement of the azimuths of both transient signals and continuous noise between 20 and 500 Hz, including diffuse wind-driven ocean noise. The lack of significant shipping or industrial noise in this region provides a rare opportunity to directly measure the properties of wind-driven noise in expanding ice-free regions. Here, we map the azimuthal directionality of the diffuse Beaufort ambient noise field as a function of frequency and location across all seven seasons. The dominant directionality of the diffuse ambient noise field varies strongly with frequency and is highly correlated with the received power spectral density. Certain directional features of the ambient noise field remained stable over seven deployment seasons, suggesting that judicious processing of the ambient noise soundscape could provide underwater navigational information in arctic waters. [Work sponsored by ONR.]

8:35

1aAO3. An analysis of the soundscape of a tidewater glacial fjord environment, LeConte Glacier, Alaska. Matthew C. Zeh (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1300 West 24th St., Apt. 212, Austin, TX 78705, mzeh@utexas.edu), Erin C. Pettit (Dept. of GeoSci., Univ. of Alaska Fairbanks, Fairbanks, AK), and Preston S. Wilson (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Glacierized fjords present a unique set of acoustic environments that are significantly louder than other ice-covered environments, with average sound pressure levels of 120 dB (re 1 μPa) with a broad peak between 1 and 3 kHz [*Geophys. Res. Lett.* **42**, 205 (2015)]. The intensity within this peak is due to the release of bubbles from compressed air-filled pores within melting glacier ice. The glacier-ocean boundary is dynamic, sensitive to fresh- and sea-water balances governed by submarine glacier melt from heat transfer between the glacier face and ocean [*J. Phys. Oceanogr.* **46** (2016)]. This heat transfer leads to regular calving events (< 200 Hz) contributing significantly to the overall sound level of the environment. Further, as icebergs melt and break apart they may remain in the fjord forming an ice mélange top boundary layer. This talk presents acoustic measurements from and analysis of the soundscape of LeConte Bay, a glacierized fjord in southeastern Alaska, studied from October 2015 to April 2016 using a 6-element vertical hydrophone array. The hydrophones recorded simultaneously at 48 kHz for 2 minutes every hour at depths ranging from 100 to 200 m approximately 1 km from LeConte Glacier. [Work supported by NDSEG Fellowship and ONR.]

8:50

1aAO4. Experimental studies of ice cracking tests in an anechoic chamber. Matthew V. Ahlrichs (Civil Eng., Univ. of Alaska, 3211 Providence Dr., Anchorage, AK 99508, matthew.ahlrichs@gmail.com), Chenhui Zhao, Marehalli Prasad (Mech. Eng., Steven's Inst. of Technol., Hoboken, NJ), James Matthews (Civil Eng., Univ. of Alaska, Anchorage, AK), Steven Opet (Elec. Eng., Stevens Inst. of Technol., Hoboken, Alaska), James Lyon (Chemical Eng., Stevens Inst. of Technol., Hoboken, NJ), Trevor Hinds (Mech. Eng., Steven's Inst. of Technol., Hoboken, NJ), and Khiana Rogers (Civil Eng., Northeastern, Boston, MA)

As the Northwest passage becomes more frequently traveled by commercial and business interests, it is becoming important to increase existing methods for safety at sea in harsh Arctic conditions. In order to identify ice-floe movements in the Arctic, acoustical signatures for ice behaviors need to be identified and analyzed. Being able to identify the acoustic signature of ice cracking in real-time will help vessels navigate the arctic by identifying moving ice-floes and monitoring present ice conditions. This study identifies a novel method for detecting thermal cracking in ice, ice fracturing, and ice shearing events in an anechoic chamber. Comparing the power spectra and identified peak frequencies against empirically collected data from previous studies, this methodology aims to recreate the arctic environment in a lab without travel to the region for baseline data collection. Furthermore, based on a literature review, little power spectra data exist. The data collected from this study will add to the creation of a baseline identification for future ice fracture studies. Observations are made on the collected data which can be used to improve navigational methods for safety at sea.

9:05

1aAO5. The sounds of submarine volcanoes. Gabrielle Tepp, Matthew Haney, John Lyons (USGS Alaska Volcano Observatory, 4230 University Dr. Ste. 100, USGS Alaska Volcano Observatory, Anchorage, AK 99508, gtepp@usgs.gov), Robert Dziak (NOAA/PMEL, Newport, OR), Joe Haxel (OSU/CIMRS, Newport, OR), Del Bohnenstiehl (Dept. of Marine, Earth & Atmospheric Sci., North Carolina State Univ., Raleigh, NC), and William Chadwick (OSU/CIMRS, Newport, OR)

When submarine volcanoes erupt, several processes can create sounds in the ocean, mostly at low frequencies <100 Hz. Explosions may occur directly in the water column, while earthquakes and other seismicity may produce seismic waves that convert into hydroacoustic waves or boundary (Scholte) waves. Volcanic sounds can propagate large distances through the SOFAR channel. During its 2014 eruption, Ahyi seamount, Northern Mariana Islands produced repetitive signals for approximately 2 weeks at a high rate. These likely explosions were widely recorded on seismometers throughout the region and on hydrophone arrays as far as Chile, ~12,000 km distant. Bogoslof volcano, a shallow submarine volcano in the Aleutian Islands, Alaska, began erupting in December 2016. Many of the detected earthquakes associated with this eruption have large amplitude hydroacoustic phases, likely Scholte waves. A few earthquake swarms were recorded as converted hydroacoustic waves by seismometers on Tanaga volcano, ~700 km away. A few eruption sequences were also detected on the Tanaga stations, one of which included a monochromatic glide, suggesting efficient transmission of energy into the water column. A single hydrophone deployed near Bogoslof several months after the eruption began may add evidence for how eruptions of submarine volcanoes contribute to the underwater soundscape.

9:20

1aAO6. Partitioning wind and ship generated sound using vertical noise coherence. Najeem Shajahan and David R. Barclay (Oceanogr., Dalhousie Univ., 1355 Oxford St., PO Box 15000, PO Box 15000, Halifax, NS B3H 4R2, Canada, nj210471@dal.ca)

Continuous ambient noise data were recorded during April and May 2016, using a four-element vertical array deployed near the continental shelf break south of Martha's Vineyard. A technique for classifying and partitioning ambient noise using the vertical noise coherence function is proposed. Time series analysis of the noise power spectrum reveals the presence wind, distant shipping, and near-field individual ship noise in the region. The noise coherence (directionality) due to wind, distant shipping, and individual ships is analytically modeled using environmental inputs such as the time varying sound speed profile and sediment properties from the measurement site, and compared with the observation. The impact of noise due to ship traffic in the region is estimated by subtracting the best-fit theoretical coherence for wind-generated noise from the measurement. Since the wind generated vertical coherence is stable and independent of source spectrum level, it can be used to quantify the relative contributions of distant shipping and wind noise to the marine environment. Additionally, the time varying vertical coherence from near-field individual ships can be used for estimating their range and speed. [Research supported by ONR.]

9:35

1aAO7. Progress toward acoustic volume attenuation tomography. Christopher M. Verlinden, Jeffery D. Tippmann, William A. Kuperman, and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu)

Experiments were carried out in the eastern Pacific and north Atlantic Oceans with the purpose of estimating volume attenuation coefficients in seawater by measuring acoustic amplitude along eigenray paths in the mid-frequency regime (3-9 kHz). A quantitative comparison of the measured attenuation coefficients in three experiment sites is presented here along with a comprehensive examination of historical measurements in these areas. The ability to make precise measurements of acoustic volume attenuation coefficients has applications in detection theory, acoustic communications, and ocean sensing. A method of inverting for properties such as temperature, salinity, and pH, which influence the attenuation of sound in seawater, is discussed.

9:50

1aAO8. Three-dimensional noise modeling. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca) and Ying-Tsong Lin (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

The ambient sound field due to wind generated surface noise and distant ship generated noise may be highly axially non-symmetric in regions with rapidly changing bathymetry. Some of this asymmetry is due to bathymetric shadowing, while in other cases horizontal refraction contributes significantly. These propagation effects can alter the omnidirectional power spectral levels, as well as the horizontal and vertical noise coherence (directionality). The special case of an idealized Gaussian submarine canyon can be described using the method of normal mode decomposition applied to a three-dimensional longitudinally invariant wave-guide. The modal decomposition is carried out in the vertical and across-canyon horizontal directions and gives a semi-analytical solution describing the three-dimensional bathymetric effects on the noise field. Reciprocal three-dimensional cylindrical co-ordinates parabolic equation (PE) and Nx2D PE sound propagation models can be used to compute the noise field in arbitrary domains. Inter-comparison of these models highlights the effect of the three-dimensional topography on the vertical coherence and mean-noise level as a function of arrival direction relative to the canyon's axis. These effects include the focusing of noise along the canyon axis and the frequency perturbation of vertical coherence minima. [Research supported by ONR.]

MONDAY MORNING, 4 DECEMBER 2017 BALCONY N, 9:00 A.M. TO 10:00 A.M.

Session 1aEAa

Engineering Acoustics: General Topics in Engineering Acoustics I

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

Contributed Papers

9:00

1aEAa1. Legendre polynomial method for linear array beam pattern synthesis. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

The detailed design tools for acoustic array sidelobe suppression and beamwidth control by Legendre polynomials technique for equally spaced point source linear acoustic array synthesis are studied and introduced. The method is compared with results obtained by conventional classical Dolph-Chebyshev synthesis technique. Numerical examples were demonstrated by a group of Legendre polynomials of different degrees of N with z_0 controlling parameters. Fourier series cosine expansion is applied to calculate element weight coefficient for great advantage when the number of sensor count in an array goes large. Naturally taped sidelobe levels within a beam pattern, and better beam efficiency, a measure of the energy percentage to the mainlobe, were both obtained by Legendre polynomial array synthesis method.

9:15

1aEAa2. Acoustic beam pattern for a fringe free laser interferometric sound sensor. Michael S. McBeth (SSC Atlantic, SSC Atlantic/NASA Langley Res. Ctr., 11 West Taylor St., M.S. 207, Hampton, VA 23681, m.s.mcbeth@ieee.org)

A fringe free laser interferometric sound sensor uses parallel, non-intersecting, modulated laser beams in a sound-filled fluid. Therefore, no interference fringes are produced or required in the fluid. Light from each laser beam is collected and combined in an envelope detector where the sound pressure signal appears as angle modulated sidebands. Since the laser beams do not intersect in the sound-filled fluid, we can use a beam separation distance that maximizes the optical path difference between the laser beams to increase sound detection sensitivity. We derive an equation that relates the optical path difference between a pair of laser beams and the peak change in the index of refraction of the fluid due a sound wave. We derive a three dimensional beam pattern and function for this sensor and compare it to that of a conventional line array.

9:30

1aEAa3. Acoustic field approximation with sparse arrays through optimization. Lane P. Miller, Stephen C. Thompson (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lpm17@psu.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

In a previous work, the objective of approximating the acoustic field from an arbitrary continuous aperture with constrained sparse transducer arrays was established. Results from this showed limited accuracy due to the constraints and called for further consideration. The aim of this research is to extend the analysis and increase the accuracy of the field approximation through multi-variable optimization. System parameters such as transducer excitation level and location within the aperture space are varied. The acoustic coupling between transducers is also taken into account. Results from the optimizations will provide the system parameters that minimize error between compared radiation patterns.

9:45

1aEAa4. Guidelines and principles for noise and vibration performance in an autonomous vehicle. Arun Mahapatra (Vehicle Eng., Ford Motor Co., 20800 Oakwood Blvd., Dearborn, MI 48124, amahapa1@ford.com)

The field of autonomous vehicles (AVs) has grown substantially in recent years, both in public interest and engineering efforts. This presents a unique challenge in the world of automotive engineering—many processes and principles are based on the presence of an engaged driver or operator, while a passenger in an AV is assumed to have a much lower level of engagement. Additionally, AVs are an entirely new, quickly evolving market that has no historical data to use as a basis for engineering decisions, including those related to the noise, vibration, and harshness (NVH) experienced by a passenger. This paper reviews previous studies focusing on the NVH experiences of automobiles, other forms of transportation (trains, planes), and recent psychoacoustic findings to generate a set of principles and guidelines for NVH development of an AV. While there is minimal real-world information available on a consumer's experience with an AV, a remarkable amount of information from other industries and environments can be applied to ensure an optimal NVH environment for a passenger of a self-driving vehicle.

MONDAY MORNING, 4 DECEMBER 2017 — BALCONY N, 10:30 A.M. TO 11:15 A.M.

Session 1aEAb

Engineering Acoustics: General Topics in Engineering Acoustics II

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

Contributed Papers

10:30

1aEAb1. Full-field laser vibrometer study of dissipation in a beam with a coupled oscillator array. Jenna Gietl, John A. Sterling (The Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064, jenna.gietl@gmail.com), Teresa J. Ryan (East Carolina Univ., Greenville, NC), Joseph F. Vignola, and Diego Turo (The Catholic Univ. of America, Washington, DC)

Coupled oscillator systems can be designed to manage the distribution of mechanical energy in vibration or acoustic systems. A specific implementation of an array composed of small damped mass-spring attachments on a primary is here referred to as a subordinate oscillator array. SOAs and their ability to absorb energy from a primary mass are of continued importance in analysis for both structural and acoustic systems. The ability to dissipate energy in a specific frequency band has many potential applications. This work will use laser Doppler vibrometry (LDV) to evaluate a beam mounted on an electromechanical shaker with and without an attached planar SOA composed of much smaller beams. This study is motivated by earlier work that indicated a high sensitivity to disorder in the form of fabrication error. The apparatus allows for the controlled introduction of various levels of disorder to quantify its effect and compare to a numerical model. We will show full-field forced response from the LDV with and without the SOA and the resulting apparent damping within the band of the SOA for various levels of disorder.

10:45

1aEAb2. Toward acoustic particle velocity sensors in air using entrained balloons: Measurements and modeling. Randall Williams, Donghwan Kim (The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg 160, Rm. 1.108, Austin, TX 78758, r.williams.06@gmail.com), Kristofer L. Gleason (Silicon Audio, Inc., Austin, TX), and Neal A. Hall (The Univ. of Texas at Austin, Austin, TX)

Acoustic particle velocity sensors for underwater applications typically work by measuring the motion of a small, neutrally buoyant body entrained in the acoustic field. Motivated by this, we have investigated the use of balloons for acoustic particle velocity measurements in air by exploring the behavior of an elastic balloon vibrating in response to an incident acoustic wave. In this presentation, we will present the results of experiments performed in an anechoic chamber, in which a pair of laser Doppler vibrometers simultaneously captured the velocities of the front and back surfaces of a Mylar balloon in an acoustic field. From phase measurements, the motion is described in terms of contributions from odd-order vibration modes (including bulk translation) and even-order vibration modes. The measured entrainment factors for the balloon at low frequencies are seen to be in good agreement with a physical model based on the scattering from an entrained, rigid sphere. This demonstrates the feasibility of using balloons for direct measurement of acoustic particle velocity in air.

11:00

1aEAb3. An experimental investigation of the nonlinear acoustic response of acoustic sense-ports. Thomas W. Teasley and David Scarborough (Aerosp. Eng., Auburn Univ., 211 Eng. Dr., Auburn, AL 36849, twt0007@auburn.edu)

Combustion instabilities continue to hinder the development of rocket engines and high-efficiency, low NOx combustion technology used in gas turbine engines. Experimental pressure measurements remain the best method to assess combustion instabilities. However, the harsh, high-temperature environment requires remotely mounting pressure sensors using sense-ports, which cause large discrepancies in measured thrust chamber acoustic pressure amplitudes. For this study, a multi-microphone impedance tube was used to investigate the nonlinear response of an acoustic sense-port. Measurements were performed for frequencies and driving amplitudes ranging from 100 Hz to 1500 Hz and 120 dB to 175 dB, respectively. Measurements were made using four different sense-port area-contraction ratios and for different extension tube lengths. The sense-port and extension tube acoustic responses were measured separately to enable the determination of the abrupt area contraction acoustic response. Measured sense-port area contraction length corrections were found to be in close agreement with the literature. The rigidly terminated sense-port extension tube exhibited linear acoustic damping. Measurements of the abrupt area contraction acoustic response revealed highly nonlinear damping even at low acoustic pressure amplitudes due to flow separation at the abrupt area contraction caused by the local acoustic velocity.

MONDAY MORNING, 4 DECEMBER 2017 STUDIO 2, 7:50 A.M. TO 12:00 NOON

Session 1aNS

Noise and Physical Acoustics: Supersonic Jet and Rocket Noise I

Caroline P. Lubert, Cochair
Mathematics & Statistics, James Madison University, 301 Dixie Ave., Harrisonburg, VA 22801

Kent L. Gee, Cochair
Brigham Young University, N243 ESC, Provo, UT 84602

Chair's Introduction—7:50

Invited Papers

7:55

1aNS1. Sixty years of launch vehicle acoustics. Caroline P. Lubert (Mathematics & Statistics, James Madison Univ., 301 Dixie Ave., Harrisonburg, VA 22801, lubertcp@jmu.edu)

On 4 October 1957 at 7.28 pm, the first artificial low Earth orbit satellite, Sputnik, was launched by the Soviet Union. Its launch ushered in a host of new scientific and technological developments, and public reaction in the United States led to the so-called "Sputnik Crisis," and the subsequent creation of NASA. A race ensued between the United States and the Soviet Union to launch satellites using carrier rockets. At this time, very little was known about the acoustics of rocket launches, and even less about acoustic suppression. Thus, in the vicinity of the rocket, acoustic levels could reach up to 200 dB during lift-off. Such extremely high fluctuating acoustic loads were a principal source of structural vibration, and this vibro-acoustic interaction critically affected correct operation of the rocket launch vehicle and its environs, including the vehicle components and supporting structures. It soon became clear that substantial savings in unexpected repairs, operating costs, and system failures could be realized by even relatively small reductions in the rocket launch noise level, and a new discipline was born. This paper presents a review of the first 60 years of launch vehicle acoustics.

8:15

1aNS2. On the acoustic near field of a solid propellant rocket. Christopher Tam (Mathematics, Florida State Univ., 1017 Academic Way, Tallahassee, FL 323064510, tam@math.fsu.edu)

Recently, Horne *et al.* presented NASA Ames measurements of the acoustic near field of a solid propellant rocket. The experiment consists of two phases: the high-burn and the low-burn phase. The main objective of this investigation is to use this set of data for the determination of the dominant components of near field rocket noise. The data consist of spectral measurements of an array of 14 near field microphones and a single far field microphone. By itself, the data are insufficient to accomplish the stated objective. We supplement the data with information provided by the two-noise source model of hot supersonic laboratory jets. The two-noise source model is supported by the existence of two similarity spectra. By applying the similarity spectra to the data of Horne *et al.* at low-burn, we are able to show that the dominant components of near field solid propellant rocket noise are the same as those found in the far field of supersonic jets. Further, the data allow the development of a model for the spatial distribution of the dominant noise components. On applying the model to the high-burn phase of the experiment, excellent agreements are found.

8:35

1aNS3. Investigation of single and twin jet interactions with plates with cut-outs. Karthikeyan Natarajan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., EAD, PB 1779, Old Airport Rd., Bangalore 560017, India, nkarthikeyan@nal.res.in) and Lakshmi Venkatakrishnan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., Bangalore, Karnataka, India)

The design and testing of the launch vehicle structure and its subsystems to withstand the lift-off acoustic loads is quite a daunting task in itself. Any effort to minimize these loads can prove to be highly beneficial as it directly influences the design, weight, and qualification of the launch vehicle components and hence the overall vehicle operating cost and time. The components of launch pad such as the launch platform and jet blast deflector are known to be the principal noise sources of the intense acoustic loads generated during lift-off. They contribute to the overall noise levels experienced by the launch vehicle by either reflecting the noise generated by the jet exhaust or by creating additional sources of noise. Earlier studies showed that the presence of cut-outs in the launch platform significantly affects the overall acoustic loads experienced by the launch vehicle. The present paper attempts to characterize the influence of cut-outs in the launch platform on the noise levels experienced by the launch vehicle, by investigating single and twin jets impinging on flat plates with and without cut-outs at varying lift-off distances. The results from acoustic measurements carried out in the near and far-field of scaled down single and twin jet launch vehicle models are discussed. The paper also attempts to relate changes in acoustic field, brought about by the different platform configurations, to the changes in flow field through flow visualization using the schlieren technique.

8:55

1aNS4. Results of subscale model acoustic tests for H3 launch vehicle. Wataru Sarae, Keita Terashima (JAXA, 2-1-1 Sengen, Ibaraki, Tsukuba 305-8505, Japan, sarae.wataru@jaxa.jp), Seiji Tsutsumi (JAXA, Sagamihara, Kanagawa, Japan), Tetsuo Hiraiwa (JAXA, Kakuda, Japan), and Hiroaki Kobayashi (JAXA, Kanagawa, Japan)

A subscale acoustic test, the H3-scaled Acoustic Reduction Experiments (HARE), was conducted to predict liftoff acoustic environments of the H3 launch vehicle currently being developed in Japan. The HARE is based on 2.5% scale H3 vehicle models, which is composed with a GOX/GH2 engine and solid rocket motors, Movable Launcher (ML) models with upper deck water injection system and Launch Pad (LP) models with deflector and lower deck water injection systems. Approximately 20 instruments measured far/near field acoustic and pressure data. Preliminary results are presented in this presentation.

9:15

1aNS5. Experimental study of the aeroacoustic interaction between two supersonic hot jets. Hadrien Lambaré (CNES, CNES Direction des lanceurs 52 rue Jacques Hillairet, Paris 75612, France, hadrien.lambare@cnes.fr)

The first stage of space launchers often use multiple engines. The supersonic jet noise at lift-off is a major source of vibrations for the launcher's equipments and payloads. In parallel with the development of the Ariane 6 launcher and its launch pad ELA4, the CNES MARTEL test bench have been improved in order to study experimentally the aeroacoustic interaction between two hot supersonic jets (mach 3, 2000K). In collaboration with the PPRIME laboratory of the University of Poitiers, acoustic and PIV measurements have been made in the free jets configurations. Further test campaigns will study the interaction of jets inside the flame duct, and their impingement on the launch table.

9:35

1aNS6. Experimental study of plate-angle effects on acoustic phenomena from a supersonic jet impinging on an inclined flat plate. Masahito Akamine, Koji Okamoto (Dept. of Adv. Energy, Graduate School of Frontier Sci., Univ. of Tokyo, 5-1-5, Kashiwanoha, Kashiwa, Chiba 277-8561, Japan, akamine@thermo.t.u-tokyo.ac.jp), Susumu Teramoto (Dept. of Aeronautics and Astronautics, Graduate School of Eng., Univ. of Tokyo, Bunkyo, Tokyo, Japan), and Seiji Tsutsumi (Aerosp. Res. and Development Directorate, Res. Unit III, Japan Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

Acoustic waves from a rocket exhaust jet cause the intense acoustic loading. Because the exhaust jet impinges on a flame deflector at liftoff of a launch vehicle, an adequate understanding of the acoustic phenomena from a supersonic impinging jet is required for prediction and reduction of the level of the acoustic loading. The previous numerical studies on a supersonic jet impinging on an inclined flat plate [e.g., Tsutsumi *et al.* AIAA 2014-0882; Nonomura *et al. AIAA J.* **54**(3), 816-827 (2016)] suggested that the plate angle has a large impact on the characteristics of the acoustic waves from the impingement region. In the present study, experiments were carried out to discuss the effect of the plate angle on this acoustic phenomenon. By applying the acoustic-triggered conditional sampling to the schlieren visualization movies, which was proposed by the authors [AIAA 2016-2930], the phenomena around the impingement region were observed in detail. The results revealed that the plate-angle variation leads to the change in the characteristics of the acoustic waves from the impingement region, such as the source locations.

9:55

1aNS7. Modeling community noise impacts from launch vehicle propulsion noise. Michael M. James and Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com)

Commercial space is an emerging and evolving market as evidenced by the vast array of launch vehicles under development as well as the growing number of active and proposed launch sites. Federal Aviation Administration regulations require all new spaceports and launch vehicles to acquire a license. Part of the application process requires an environmental review to address the potential noise impacts to the environment and local communities. Accurate predictions of noise exposure from launch vehicles require models that have been validated over a range of vehicle types, operations, and atmospheric conditions. A high-fidelity launch vehicle simulation model, RUMBLE, has been developed to predict community noise exposure from spaceport launch, reentry, and static rocket operations. RUMBLE implements industry standard modeling practices to efficiently compute sound pressure level time histories, maximum levels, and sound exposure levels using the vehicle's engine parameters and geo-referenced source/receiver definitions. An overview of RUMBLE's underlying physics and the results of initial validation efforts (using multiple full-scale launch measurements) will be presented.

10:15–10:30 Break

10:30

1aNS8. Large eddy simulations of launcher lift-off noise and comparisons to experiments on model flame trenches. Julien Troyes (ONERA, Châtillon Cedex, France), Francois Vuillot, Adrien Langenais (ONERA, 29 Ave. de la Div. Leclerc, CHATILLON F92322, France, francois.vuillot@onera.fr), Hadrien Lambaré, and Pascal Noir (CNES, PARIS, France)

During the lift-off phase of a space launcher, rocket motors generate harsh acoustic environment that is a concern for the payload and surrounding structures. Hot supersonic jets contribute to the emitted noise from both their own noise production mechanisms and their interactions with launch pad components, such as the launch table and flame trenches. The present work describes the results of computations performed by ONERA to predict the lift-off noise from reduced scale models of a flame trench. The results include both unsteady flow solution inside the flame trench and the computed noise on near and far field microphone arrays. Numerical computations involve two in-house codes: the flow solver CEDRE, used in LES mode to accurately predict the noise sources, and the acoustic code KIM to reconstruct the far field noise, thanks to an integral Ffowcs Williams and Hawkings porous surface approach. The computational model exactly reproduces the flame trench configurations used in a test campaign carried out by CNES at the MARTEL facility. Results are discussed and compared with experimental acoustic measurements on 48 microphones. Overall, the numerical results reproduce the acoustic measurements within 3 dB. To further improve these results, work is ongoing on acoustic nonlinear effects.

10:50

1aNS9. Refinements in RANS-based noise prediction methodology for complex high-speed jets. Dimitri Papamoschou and Andres Adam (Mech. and Aerosp. Eng., Univ. of California, Irvine, 4200 Eng. Gateway, Irvine, CA 92697-3975, dpapamos@uci.edu)

Recent efforts on RANS-based modeling of the noise reduction from multi-stream, high-speed jets have underscored the importance of flow properties on the outer surface of peak Reynolds stress (OSPS). In a time-averaged sense, the OSPS is expected to represent the locus of the most energetic eddies in contact with the ambient fluid. The acoustic Mach number on the OSPS is a proxy for the convective Mach number of those eddies, thus has strong impact on the modeling of the noise source and its suppression when the jet plume is distorted into an asymmetric shape. Therefore, accurate detection of the OSPS is a critical ingredient in the modeling. This is complicated by the fact that, in asymmetric jets, the Reynolds stress distribution can be highly irregular and detection of its maximum along a radial line can become problematic. The talk will present advanced algorithms for the detection of the OSPS and resulting improvements in the prediction of noise reduction.

11:10

1aNS10. Validation of aero-vibro acoustic simulation technique using experimental data of simplified fairing model. Seiji Tsutsumi, Shinichi Maruyama (JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Wataru Sarae, Keita Terashima (JAXA, Ibaraki, Tsukuba, Japan), Tetsuo Hiraiwa (JAXA, Kakuda, Japan), and Tatsuya Ishii (JAXA, Tokyo, Japan)

To predict harmful acoustic loading observed at lift-off of the launch vehicle, aero-vibro acoustic simulation technique is developed. High-fidelity large-eddy simulation with computational aeroacoustics based on the full Euler equations in time domain are employed to predict generation of the acoustic waves and their propagation to the fairing. Coupled vibro-acoustic analysis based on finite element method are applied to compute the transmitted acoustic wave into the fairing both in the time and frequency domains. Acoustic measurement of a simplified fairing model with a subscale liquid rocket engine is conducted, and validation study of the present technique is performed. Reasonable agreement is obtained for peaks of the acoustic spectrum taken inside the fairing model. Such peaks are found to be related to the internal acoustic modes and ring modes of the faring structure. However, further study is required to obtain quantitative agreement.

Contributed Papers

11:30

1aNS11. Jet noise prediction via coupling large eddy simulation and stochastic modeling. Joshua D. Blake (MS State Univ., MS State University HPC Bldg., Office 365, MS State, MS 39762, jdb621@msstate.edu), VASILEIOS SASSANIS (MS State Univ., Starkville, MS), David Thompson (MS State Univ., MS State, MS), Adrian Sescu (MS State Univ., Starkville, MS), and Yuji Hattori (Tohoku Univ., Sendai, Japan)

A novel method for efficient and accurate jet noise prediction is developed and tested in the framework of coupled LES and stochastic noise modeling. In the proposed method, the low frequency range of the acoustic spectrum, which corresponds to large turbulent structures, is resolved with an implicit LES model by employing higher-order spatial and temporal discretizations. The high frequency range, which corresponds to the fine-scale turbulence, is modeled via a stochastic broadband noise generation model. The inputs to the stochastic model are represented by statistics from an axisymmetric RANS jet simulation. The smaller synthetic scales are convected by larger scales, accounting for the effects of sweeping from the larger LES-resolved turbulent scales. The farfield acoustic data is obtained using either the Linearized Euler Equations, or an acoustic analogy based on the Ffowcs-Williams Hawkings method. The method is evaluated on cold and heated jets at different Reynolds numbers.

11:45

1aNS12. Validation of a hybrid nonlinear approach for jet noise prediction and characterization. Vasileios Sassanis, Joshua Blake, Adrian Sescu (Dept. of Aerosp. Eng., MS State Univ., 2041 Blackjack RD, 47 Scenic Pass, Starkville, MS 39759, vs501@msstate.edu), Eric M. Collins (Ctr. for Adv. Vehicular Systems, Starkville, MS), Robert E. Harris (CFDRC, Huntsville, AL), and Edward A. Luke (Comput. Sci. and Eng., MS State Univ., Starkville, MS)

In this study, a hybrid approach for non-linear jet noise predictions in complex environments is presented and validated. The method differs from traditional approaches in that interactions of the jet with the surrounding structures as well as non-linear disturbances propagating over large

distances are taken into consideration in order to quantify their effects on sound generation and propagation. The noise sources founded in the jet plume and the near-field are first computed using the full Navier-Stokes equations. The variables of interest are then interpolated into a second domain. After penalized, they are used as source terms in the non-linear Euler equations to calculate the sound propagation. The interpolation and penalization steps are performed using a buffer region designed by the principles of sponge layers. The effective one-way communication between the two domains and the capabilities of the buffer region to transfer the data from the NS to the Euler domain without any loss of detail is demonstrated. Results from two- and three-dimensional jets both in free space and interacting with solid obstacles are presented. Comparisons with DNC and experimental data in terms of sound pressure level spectra are in good agreement with the ones calculated by our method.

MONDAY MORNING, 4 DECEMBER 2017 BALCONY L, 8:55 A.M. TO 12:00 NOON

Session 1aSA

Structural Acoustics and Vibration and ASA Committee on Standards: Standards in Structural Acoustics and Vibration

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Chair's Introduction—8:55

Invited Papers

9:00

1aSA1. A review of ASTM International standards relating to impact sound transmission in buildings. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Impact sound transmission in buildings is an important issue, especially in today's multifamily marketplace where the trend to replace carpet with hard surface flooring is becoming more popular every year. This presentation will discuss the many changes that have been made over the years to the laboratory and field test methods for evaluating impact noise in buildings, including new standards under development.

9:20

1aSA2. Acoustical Society of America Standards in Mechanical Shock and Vibration. Charles F. Gaumond (Standards, Acoust. Society of America, 14809 Reserve Rd., Accokeek, MD 20607, charles.f.gaumond.acoustics@gmail.com) and Neil B. Stremmel (Standards, Acoust. Society of America, Melville, NY)

The Acoustical Society of America maintains standards in several areas. We present the areas covered by ANSI/ASA S2: mechanical shock and vibration and ISO TC 108: mechanical shock, vibration, and condition monitoring. A brief description of each standard is given, with the overlap between ANSI and ISO standards. The work in each area is handled by committees and working groups that represent a diversity of opinions, which is crucial to creating a standard based on consensus of all interested parties. We also present the requirements for official participation in standards, that includes the rights to propose new standards, review and modify old standards, and vote. We present the reasonable fee structures for various types of organizations. Contact information for each committee and working group as well as the location of the information on the ASA Standards website are presented.

9:40

1aSA3. Development of a rating of the improvement of high-frequency impact noise. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

The authors have introduced a family of new impact noise isolation ratings for evaluating high-frequency impact sound, which encompasses the frequency range of 400-3150 Hz. The advantages of this rating include better correlation with subjective reaction and improved rank-ordering of finish flooring and sound mats. A natural extension of this metric is to define a rating of the improvement in impact isolation analogous to ΔIIC per ASTM E2179. The resulting metric ΔHIIC evaluates the improvement in high-frequency impact isolation due to floor coverings as measured in the laboratory. The research indicates that ΔHIIC predicts the result due to floor coverings considerably better than ΔIIC. Potential applications of the rating for the design and evaluation of floor-ceiling assemblies is presented.

10:00

1aSA4. Considerations for the evaluation of impact sound from heavy impact sources in adjacent spaces. William C. Eaton and Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, chris_e1@sacnc.com)

The ability to limit a sound heard by a neighboring tenant in a mixed-use facility can be achieved by understanding how to control the source of the sound or affect its path. It is also important to evaluate or measure the sound in a way that considers what a tenant might hear. These aspects will be considered when studying impact sound resulting from the drop of heavy impact sources in adjacent spaces.

10:20–10:40 Break

10:40

1aSA5. Method for evaluating and predicting noise from hard, heavy impact sources. John LoVerde, David W. Dong, Richard Silva, and Antonella Bevilacqua (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Noise and vibration from activity in fitness facilities, in particular, drops of hard heavy weights, is a common source of disturbance and complaint in residential, commercial, and mixed-use building types. Mitigating systems exist, including specially designed rubber flooring tiles and products for architectural vibration isolation systems, but in the United States, there is no standardized method for evaluating the reduction in noise or vibration provided by these products and systems. Quantitative and comparable data of the effectiveness of these products are therefore lacking. The authors previously reported (Internoise 2015, Noise-Con 2016, ASA Boston 2017) a preliminary test method to evaluate athletic tile flooring with heavy weight drops, based on the reduction in floor vibration (delta-L_V) achieved due to the insertion of the products. Preliminary results indicated that the delta-L_V measurement adequately described the impact reduction due to athletic tile, over a certain frequency range, reasonably independent of structure. This paper continues the research and further refines the measurement method, verifies its use for designing fitness facilities, and examines additional parameters that may affect measuring and evaluating hard, heavy weight impacts.

11:00

1aSA6. Non-standard uses of the mechanical tapping machine in field measurements. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

The mechanical tapping machine creates a continuous series of floor impacts which generate a steady-state sound field when measured indoors. Because each individual hammer has the same weight and free-falls from the same distance, the machine creates a calibrated and steady impulsive force into the floor structure. This presentation will illustrate potential future test methods for assessing sound transmission in buildings with examples to evaluate structure-borne sound transmission into structurally isolated anechoic and reverberation rooms.

11:20

1aSA7. Methods for measuring the vibratory response of the ground. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

A draft American National Standards Institute (ANSI) standard on "Methods for Measuring the Vibratory Response of the Ground" is nearing completion for review. This standard is being developed in parallel with a separate draft standard on "Methods for the Prediction of Ground Vibration from Rail Transportation Systems." The intent of both documents is to standardize methods that were initially developed more than 30 years ago and adopted by the Federal Transit Administration. This paper will outline the topics in the draft standard, the measurement, and analysis techniques typically associated with the methods described, and how the draft standard relates to existing standards pertaining to measuring transfer mobility.

11:40

1aSA8. Methods for calculating flanking noise transmission. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Airborne sound isolation between spaces is determined not only by the direct sound transmission through the separating assembly, but also flanking transmission, which is the structureborne transmission of vibration due to acoustical excitation in the source room. In many cases, the flanking paths dominate, such as where the separating assembly has a high level of sound transmission loss, and the ability to calculate these paths is therefore crucial if the isolation is to be accurately predicted. Methods for calculating impact noise have been developed in Europe [standard EN ISO 12354] and are currently in use in Europe and Canada; however, the methods are not commonly known or used in the United States. The authors report on use of these methods in calculating flanking noise transmission in recent projects.

Session 1aSP

Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics: Source Tracking with Microphone/Hydrophone Arrays I

Kainam Thomas Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, DE 605, Hung Hom KLN, Hong Kong

Siu Kit Lau, Cochair
Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Drive, Singapore 117566, Singapore

Invited Papers

9:00

1aSP1. Weighted coherent processing on sparse volumetric vector sensor arrays. Brendan Nichols, James S. Martin (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), Christopher M. Verlinden (Phys., U.S. Coast Guard Acad., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A network of drifting sensors, such as vector sensors mounted to freely drifting buoys, can be used as an array for locating acoustic sources underwater. Localizing a source using traditional coherent processing methods has been improved through arbitrary selection of element-wise weights of the covariance matrix [Nichols and Sabra, JASA 2015, Vol. 138]. However, the selection of weights offers an opportunity to optimize localization performance measures such as accuracy or precision. Here, the performance of source localization is compared between optimal weightings and traditional weightings for both simulated and at-sea data collected from a freely-drifting vector sensor array deployed in the Long Island Sound.

9:20

1aSP2. Tracking aircraft flybys using a microphone array. R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu), Stephen M. Tenney, and John Noble (Acoust., US Army Res. Lab, Adelphi, MD)

Ferguson and Quinn (JASA 96(2), 1994) develop equations to describe the time-frequency characteristics of an acoustic signal radiated by an aircraft and received by a stationary microphone. These equations enable construction of curves in time-frequency space whose shape depends upon several parameters, namely, aircraft speed and height above the microphone. We have utilized Ferguson and Quinn's equations to track an aircraft fly by under very low signal-to-noise conditions using six microphones which are spaced too far apart to form beams. We make use of signal coherence across the array, and we develop a time-frequency "matched filter" with replicas that are based upon aircraft speed, height above ground, and closest point of approach distance.

9:40

1aSP3. Source tracking with linear and circular compact vector sensor arrays. Berke M. Gur (Mechatronics Eng., Bahcesehir Univ., Istanbul, Turkey), Tuncay Akal, Abdurrahman Uslu, and Melih Yildirim (SUASIS, Karanfil Sok No:1 Masukiye, Kartepe, Kocaeli 41295, Turkey, tuakal@suasis.com)

Vector sensors are directional acoustic sensors that can make collocated measurements of both the acoustic pressure and the particle motion (in general velocity or acceleration). Combining the particle acceleration or velocity measurements with pressure, it is possible to estimate the intensity of the acoustic field, which in turn is related to the direction of the net acoustic energy propagation. Recently, several novel beamforming and array processing methods have been proposed that enable the development of compact linear and circular vector sensor arrays with inter-sensor spacing much less than the traditional spacing of one-half the design wavelength. These methods, albeit differing in implementation, both rely on the extraction and processing of the so-called "acoustic modes" of the sound field and have shown to be successful in estimating source direction relative to the array in a 2-D setting. The work described here, builds on previous results and extends the direction-of-arrival estimation methods to source tracking. Several algorithms developed for this purpose and implemented on both array types are introduced. The proposed approaches are experimentally validated using air-borne and underwater sources for compact pressure and 1-D vector sensor arrays.

10:00

1aSP4. Tracking time varying multipath phase at very low signal to noise ratios. Paul J. Gendron, Hanna Desiltes (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu), and Jacob L. Silva (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Diverse acoustic environments support multipath propagation. Estimating the arrival gains allows receiver structures to coherently combine acoustic energy for improved reception. In the case of mobile acoustic sources and receivers or dynamic boundaries, the amplitudes and phases of those arrivals are time varying such that estimation is made more difficult and typically entails some kind of phase tracking loop. This can be challenging as trackers are often formed as recursions in time employing previous observations to predict the phase at present, updating each phase estimate with the most recent observation. At extremely low SNR, recursive tracking can suffer loss of lock. Since a great deal of information about the instantaneous phase is contained in observations before and after it is judicious to seek estimators that exploit all of the observations in the signalling interval to infer the time varying response phase. Presented here is a scheme for simultaneous time varying phase estimation from the reception of a long duration, high time-bandwidth product transmission. In this approach, the phase process is derived from a minimum mean square error estimate of the sparse acoustic time varying response under an assumed sparse mixture model prior over Doppler and frequency. Demonstrations in shallow water at less than −12 dB received signal to noise ratio are presented.

10:20–10:35 Break

10:35

1aSP5. Detection, localization, and classification of multiple ocean vehicles over continental-shelf regions with passive ocean acoustic waveguide remote sensing. Chenyang Zhu (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, zhu.che@husky.neu.edu), Nada Saghir (Lawrence Technolog. Univ., Southfield, MI), Haoqing Li (Northeastern Univ., Boston, MA), WEI HUANG (Northeastern Univ., Malden, MA), Olav Rune Godø (Inst. of Marine Res., Bergen, Norway), and Purnima R. Makris (Northeastern Univ., Boston, MA)

Multiple ocean vehicles, including both surface ships and unknown submerged vehicles, can be simultaneously monitored over instantaneous continental-shelf scale regions via passive ocean acoustic waveguide remote sensing (POAWRS) by employing a large-aperture densely sampled coherent hydrophone array system. Here, the approach is demonstrated for multiple merchant ships present in the Norwegian Sea in 2014. The sounds radiated underwater by ocean vehicles, dominated by narrowband tonals and cyclostationary signals, are detected in the beamformed spectrograms. Coherent beamforming of the receiver array data significantly enhances the signal-to-noise ratio of the ship-radiated signals enabling detection of ocean vehicles roughly two orders of magnitude more distant in range than a single hydrophone. The estimated bearing-time trajectory of a sequence of detections are employed to determine the horizontal location of each vehicle using the moving array triangulation technique. The estimated locations are verified by comparison with historical database of merchant ships present in the region. Several time-frequency characteristics are extracted from the detected signals and used to classify and track the ships.

10:55

1aSP6. Echolocation and flight strategies of aerial-feeding bats during natural foraging. Emyo Fujioka (Organization for Res. Initiatives and Development, Doshisha Univ., 1-3 Tatara-miyakodani, Kyotanabe, Kyoto 610-0321, Japan, emyo.fujioka@gmail.com), Fumiya Hamai, Miwa Sumiya, Kazuya Motoi (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), Dai Fukui (Graduate School of Agricultural and Life Sci., The Univ. of Tokyo, Hokkaido, Japan), Kohta I. Kobayasi, and Shizuko Hiryu (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan)

Aerial-feeding bats actively emit sonar sounds and capture large amounts of airborne insects a night. Microphone-array system allows us to know not only the positions where the bat emits sonar sounds (i.e., 3-D flight path) but how the bats dynamically control the acoustical field of view during searching and approaching target-prey. Here, we show echolocation strategy of bats during natural foraging revealed by the large-scale microphone-array system which covered the horizontal area of approximately 20 m × 20 m. *Pipistrellus abramus* was found to expand the width of their sonar beams in both horizontal and vertical planes just before the prey-capture. Since the bats emit echolocation pulses at a high rate (i.e., feeding buzz) just before capturing, the capture positions can additionally be measured. Recently, we have investigated the relationship between flight patterns, capture positions, and foraging efficiency of *Myotis macrodactylus* during natural foraging above the pond. Further investigation from the viewpoint of the optimal foraging would reveal echolocation and flight strategies of the bats to efficiently search and approach prey items. [This research was supported by a Grant-in-Aid for Young Scientists (B) and Scientific Research on Innovative Areas of JSPS, and the JST PRESTO program.]

11:15

1aSP7. Adaptive cubature Kalman filtering for distant speech tracking using a circular microphone array. Xiang Pan, Yue Bao, Yiting Zhu, Zefeng Cheng, and T. C. Yang (School of Information and Electron. Eng., Zhejiang Univ., Zhe Rd. 38, Hangzhou, Zhejiang 310027, China, panxiang@zju.edu.cn)

A joint processing framework for distant speech tracking is proposed based on combination of an adaptive cubature Kalman Filter with a 80-element circular microphone array. The deconvolved convention beamforming (DCB) [T. C. Yang, 10.1109/JOE.2017.2680818] is carried out over the distant speech data for speech enhancement and bearing estimate. DCB provides superdirective beams and offers the same robustness as conventional beamforming. With the estimated speech bearing, a "current" statistical (CS) motion model is combined with a cubature Kalman filter to track the speech of interest. The CS model assumes a modified Rayleigh distribution for the acceleration probability density (of the moving source) whose mean value is the current acceleration. Based on this model, we utilize a zoom factor achieved by computing the norm of the tracking innovation to adjust the dynamic state model adaptively. Both numerical simulation and the outdoor experimental results show that the proposed framework in this paper can effectively track a maneuvering speech source.

Session 1aUW

Underwater Acoustics: Underwater Acoustic Scattering and Reverberation

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Edward Richards, Cochair
Scripps Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093

Contributed Papers

8:15

1aUW1. Coherence in shallow water reverberation. Dajun Tang (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Coherent structure of reverberation intensity versus time and depth is observed during TREX13, and the structure resembles that of sound propagation versus range and depth. The experiment was conducted in shallow water approximately 20 m depth with both source and vertical receiving arrays moored, so the ping-to-ping incoherence in reverberation is believed to be caused by time-dependence of water column properties, especially rough sea surfaces. To understand the observed coherence, simulation of reverberation field is performed, where reverberation pressure, rather intensity, is calculated so phase coherence is included. Possibility of inferring sound propagation information from measured reverberation is discussed. [Work supported by the US Office of Naval Research.]

8:30

1aUW2. Modelling broadband scatter from periodic sea surfaces. Edward Richards, William S. Hodgkiss, and Hee-Chun Song (Scripps Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, edwardlrichards@ucsd.edu)

Surface waves are the primary source of short time-scale variation of the channel impulse response in underwater transmission scenarios when source and receiver positions are fixed. This study proposes an exact method for modeling the single surface bounce path of the channel impulse response, which contains most of short time-scale effect. This method is designed for one dimensional, periodic wave profiles, and solves the scattering problem as a series of motionless surfaces. To create a time series for each surface, the proposed method solves the Helmholtz equation at many frequencies, performing three steps at each frequency. First, the point source is decomposed into a series of plane waves, each of which scatter from the surface at a finite number of Bragg angles. Second, the amplitudes of these discrete scattered plane waves are found with an exact method. Finally, the surface response for a three-dimensional point source is recovered with a plane wave synthesis of these scattered plane waves. The results of this study are intended as a reference to inform the interpretation of surface scattering results from more common underwater acoustic models.

8:45

1aUW3. Delay-Doppler characteristics of surface interacting underwater sound. Sean Walstead (Naval Undersea Warfare Ctr. (NUWC), 1176 Howell St., Newport, RI 02841, sean.walstead@navy.mil)

The interaction of underwater sound with the ocean surface is investigated. Laboratory measurements of the delay-Doppler structure of the surface multipath at a transmission frequency of 300 kHz are compared to physics based analytic predictions. Channel impulse responses and channel scattering functions are measured when the surface is characterized by wind generated roughness, swell-like roughness, and a mix of wind and swell. The distribution of surface scattered energy across delay time and Doppler shift is presented as a function of wind speed ranging from 0.5 m/s to 11.0 m/s. The effect of wave shape and nominal out-of-plane grazing angle on the delay time spread and Doppler spread of the surface multipath are also considered. Proper characterization of surface interacting acoustic energy has direct application to the improved performance of shallow water and near surface underwater acoustic communications systems.

9:00

1aUW4. Machine learning the acoustic impulse response of a rough surface. Richard S. Keiffer and Steven Dennis (Acoust. Div., Naval Res. Lab., Bldg. 1005, Rm. C-2, Stennis Space Ctr., MS 39529, richard.keiffer@nrlssc.navy.mil)

Due to time constraints imposed by operational considerations, reverberation models used by the Navy split the computational work by modeling the propagation and the scattering separately. The scattering module is typically a "look-up" table linking the incident sound field at any point on the boundary to a pre-computed statistic of the scattering. There are at least two significant drawbacks to this standard modeling approach. First, each acoustic interaction with the boundary generates a unique realization of a stochastic process, and consequently, no single statistic of the scattering always can be justified and the correct modeling of the statistics of reverberation time series must acknowledge this event-to-event variation. Second, pre-computing the scattering based on an assumed statistical description of the unresolved boundary irregularities without consideration of the resolved boundary irregularities prohibits the coherent inclusion in that scattering calculation of the effects that the resolved boundary irregularities have on the scattering. Machine-learned models for the boundary scattering hold the promise of addressing both of these issues. The purpose of this talk is to explore these issues and the prospect of developing a machine-learned model for the acoustic impulse response of a multi-scale rough ocean boundary. [This work was supported by the U.S. Office of Naval Research.]

9:15

1aUW5. Modal analysis of split-step Fourier parabolic equation solutions in the presence of rough surface scattering. Mustafa Aslan (Turkish Naval Acad., Istanbul, Turkey) and Kevin B. Smith (Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu)

Determining accurate solutions of acoustic propagation in the presence of rough surface scattering is of significant interest to the research community. As such, there are various approaches to model the effects of rough surface scattering. Typically, the models used assume an idealized pressure release condition at the sea surface boundary. This boundary condition can easily be accommodated through a variety of modeling techniques for flat

surfaces, but it is necessary to introduce additional complex methods in numerical models for rough surfaces. Parabolic equation (PE) models utilizing split-step Fourier (SSF) algorithms have been employed to treat the rough surface displacements. These include methods such as the field transformational technique (FFT), and direct modeling of the physical water/air interface discontinuity. Previous work highlighted phase errors in SSF-based models with large density discontinuities at the interfaces, which were minimized by employing a hybrid split-step Fourier/finite-difference approach. However, such phase errors were largely absent in the presence of rough surface scattering. In this work, the PE solutions are decomposed into normal modes in order to determine which modes dominate the phase error in the presence of flat surfaces, and to confirm that these modes are highly scattered in the presence of rough surfaces.

9:30

1aUW6. An autonomous, bottom-mounted sonar for measurement of mid-frequency reverberation. Brian T. Hefner, Dajun Tang, and Taebo Shim (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

While bottom reverberation is typically dominant over surface reverberation at mid-frequencies in shallow water, sea surface roughness has a significant impact on transmission loss and hence on reverberation. Measurements of this effect can be difficult with a system deployed from a moving ship due to both the system motion and the constraints of operating the ship safely in the rough seas of interest. To overcome these difficulties, the Autonomous Reverberation Measurement System (ARMS) has been developed under an ONR-sponsored DURIP. This system is a benthic lander with a directional source and receive array mounted on a rotation stage. Powered by batteries and programmed prior to deployment, the system can measure reverberation from 2-6 kHz as a function of direction autonomously for up to 3 months. This system was deployed in the spring of 2017 as part of an experiment conducted off Geoje Island, Republic of Korea. During this experiment, the ARMS was deployed in 28 m of water and measured reverberation in a muddy sand environment with a range-dependent water depth that was punctuated with rock outcroppings on the seafloor. [Work supported by the U.S. Office of Naval Research.]

9:45

1aUW7. Experiments on the sound transmission at the water-air interface for different source-interface distances. Daniel Wehner and Martin Landrø (Dept. of GeoSci. and Petroleum, Norwegian Univ. of Sci. and Technol., S.P. Andersens Veg 15a, Trondheim, Sør-Trøndelag 7031, Norway, daniel.wehner@ntnu.no)

The water-air interface is a nearly perfect reflecting boundary for acoustic waves due to the high impedance contrast between the two media. In many marine applications, ranging from geophysical measurements to bioacoustics, the sea surface has an important impact. A rough sea surface topography leads to complex scattering and interference for acoustic frequencies with similar wavelength or higher than the wavelength of the interface. In geophysical applications and infrasound the frequencies of interest often have larger wavelengths. At the same time, the source within these applications is mostly located close to the interface with respect to its wavelength. In this case, an increased transmission could be expected as experiments with acoustic transducers demonstrate [D. C. Calvo *et al.*, *J. Acoust. Soc. Am.* **134**, 3403-3408 (2013)]. We design two experiments with sources close to the interface while changing the distance between the source and the water-air surface. For the first experiment, a water gun, which creates large cavities, is placed in water. Second, a signal gun, as used in athletics, is positioned in air. The acoustic transmission from both sides is measured and investigated. We find an increasing transmission coefficient for lower frequencies and decreasing distance between source and interface.

10:00–10:15 Break

10:15

1aUW8. Progress on applying the small-slope approximation to layered seafloors. Darrell Jackson (Appl. Phys., Univ. of Washington, 1013 NE 40th St, Seattle, WA 98105, drj@apl.washington.edu)

The small-slope approximation (SSA) for scattering by rough interfaces is an attractive alternative to the classic small-perturbation and Kirchhoff approximations, as it combines the best aspects of both and may be more accurate over all angular ranges. While SSA has most frequently been applied to impenetrable interfaces and interfaces bounding homogeneous half spaces, it has also been developed for layered media. There is no unique extension of SSA to layered media, and three different methods have appeared in the acoustics and electromagnetics literature. These methods will be reviewed and a new, possibly superior, method will be introduced. This method arises naturally when SSA is expressed in coordinate space rather than k-space. The various methods will be compared using numerical examples. [Work supported by ONR.]

10:30

1aUW9. Data analysis and modeling of broadband acoustic propagation perpendicular to internal wave fronts and sand dune crests in the South China Sea. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu), A. Y. Chang (National Sun Yat-Sen Univ., Kaohsiung, Taiwan), Chi-Fang Chen (National Taiwan Univ., Taipei, Taiwan), Ching-Sang Chiu (Oceanogr., Naval Postgrad. School, Monterey, CA), Linus Chiu (National Sun Yat-Sen Univ., Kaohsiung, Taiwan), Chis Miller (Oceanogr., Naval Postgrad. School, Monterey, CA), Steven R. Ramp (Soliton Ocean Services, Carmel Valley, California), Ruey-Chang Wei (National Sun Yat-Sen Univ., Kaohsiung City, Taiwan), and Y. J. Yang (National Taiwan Univ., Taipei, Taiwan)

Very large subaqueous sand dunes were discovered on the upper continental slope of the northeastern South China Sea (SCS) in the spring of 2007 during an ONR 322OA-funded field experiment which was designed to study the large transbasin internal solitary waves (ISW) that are generated by tidal forcing in the Luzon Strait. These internal waves and sand dunes are important acoustical features, as it is expected that they will cause significant anomalies in the acoustical field. In the spring of 2014, a broadband source was deployed to transmit 850-1150 Hz LFM signals to receivers on a mooring in the center of the sand dune field. The acoustic transect was oriented perpendicular to the dune crests, ISW fronts and isobaths. Data analysis and extended modeling are presented to quantify the degree to which these features impact the propagation of broadband signals in the 100-2000 Hz band as a function of source depth and frequency.

10:45

1aUW10. Measurement of mid-frequency acoustic backscattering from the sandy bottom at 6–24 kHz in the Yellow Sea. Guangming Kan (Marine Geology and Geophys., First Inst. of Oceanogr., Rm. 109, Main Bldg., 6th Xianxialing Rd., Laoshan District, Qinfdao, Shandong 266061, China, kgming135@fio.org.cn), Baohua Liu, Zhiguo Yang, Shengqi Yu, Kaiben Yu (National Deep Sea Ctr., Qingdao, Shandong, China), and Yanliang Pei (Marine Geology and Geophys., First Inst. of Oceanogr., Qingdao, Shandong, China)

In a typical sandy bottom area of the South Yellow Sea, measurement of acoustic bottom backscattering strength within a frequency range of 6–24 kHz was conducted using omnidirectional sources and omnidirectional receiving hydrophones. In this experiment, with interference from scattering off the sea surface being avoided and the far-field condition being satisfied, we obtained acoustic bottom backscattering strength values ranging from -31 to -17 dB within a grazing angle range of 18°–80°. In the effective grazing angle range, the acoustic scattering strength generally increases with the increase in the grazing angle, but the variation trends were different in different frequency bands, which reflects different scattering mechanisms. The frequency dependence of the acoustic backscattering strength is characterized by a segmented correlation. In the frequency band of 6–11 kHz, the scattering strength is positively correlated with the frequency, and the average slope of the linear correlation is about 0.83 dB/octave; in the frequency band of 12–24 kHz, the scattering strength generally exhibits a negative correlation with the frequency, and the slope of the linear correlation is about -0.42 dB/octave.

11:00

1aUW11. Time-domain acoustic scattering by elastic objects using finite element analysis. Blake Simon (Appl. Res. Labs., Univ. of Texas, 1400 Briarcliff Blvd., Austin, TX 78723, blakesimon8@gmail.com), Aaron M. Gunderson (Appl. Res. Labs., Univ. of Texas, Pullman, Washington), and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, Austin, TX)

Finite element analysis provides an accurate way of calculating acoustic scattering from underwater objects. The method provides an exact solution to the Helmholtz equation to the order of the discretization. Although commercial finite element software is capable of solving fully 3D scattering problems, it has been previously shown by Zampolli *et al*. [*J. Acoust. Soc. Am.* **122**, 1472–1485 (2007)] that 3D axisymmetric targets can be solved more efficiently using 2D geometry. This method uses an axial wavenumber decomposition technique, which simulates an incident plane wave from an off-axis direction. In this study, results from this analysis are converted from the frequency domain to the time domain using Fourier synthesis. Elastic spheres and cylinders are considered because the analytical solutions for scattering by these targets are known and can be used to verify the finite element results. The success of this model to simulate time domain scattering will inform the applicability of finite element analysis for more complex targets. [Work supported by ONR, Ocean Acoustics.]

11:15

1aUW12. Late backscattering enhancement from a rubber spherical shell associated with waveguide coupling and propagation. Aaron M. Gunderson (Appl. Res. Labs., Univ. of Texas at Austin, Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758, aaron.gunderson01@gmail.com), Timothy Daniel, Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA), and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Acoustic scattering from rubber targets has received modest attention in the past due in part to rubber's subsonic sound speeds. In particular, slow shear waves in rubber get rapidly attenuated, giving rubber the fluid-like property of negligible shear coupling. This makes rubber a material of interest for coating or cloaking underwater objects and vehicles. In this study, backscatter from a rubber spherical shell in water is considered experimentally and through such models as partial wave series, finite elements, and waveguide normal mode analysis. The target is a commercially available American handball of unspecified rubber composition. Experimental and modeled results exhibit the importance of a strong, slightly subsonic late echo, which is demonstrated through path timing models and frequency analysis to be due to waveguide propagation through the shell wall. The various models use wave speeds for the rubber determined experimentally, allowing for determination of the phase and group velocities of the lowest order waveguide mode within the shell. These are found through normal mode analysis for both flat and curved waveguides, and again through Sommerfeld-Watson theory. Frequency domain results show how the waveguide path interferes with reflective paths from the shell in a frequency-dependent manner. [Work supported by ONR.]

11:30

1aUW13. On target excitation by modulated radiation pressure. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com), Ivars P. Kirsteins (NUIC, Newport, RI), Philip L. Marston, and Timothy Daniel (Phys., WSU, Pullman, WA)

Modulated radiation pressure (MRP), which can be produced by modulating an ultrasound carrier beam with a low-frequency signal provides the capability to interrogate elastic objects in a novel manner: The high-frequency carrier beam enables surgical investigation of target of interest, while the low-frequency modulation signal "shakes" the target to extract useful physical information. By sweeping the modulation frequency, target resonances can be identified, and by physically scanning the target at a modulation frequency that is commensurate with one of its resonant frequencies, the corresponding mode shape can be extracted. Up until now, the application of this technique has been limited to medical ultrasound, where it has been used to look for tumors and kidney stones among other things. The other applications of this technique have been particle trapping and non-contact manipulations. In this presentation, we will show results from experiments conducted at Washington State University involving scaled targets, complemented by finite element modeling results. We will discuss coupling between two targets near each other, while one of them is excited by MRP. We will also discuss how the radiated field from a target excited by MRP scales with its size. [Work supported by ONR.]

Session 1pAA

Architectural Acoustics and Psychological and Physiological Acoustics: Perceived Diffuseness I

Jin Yong Jeon, Cochair
Department of Architectural Engineering, Hanyang University, Seoul 133-791, South Korea

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Review of the perception of diffusive surfaces in architectural spaces. Peter D'Antonio, Jeffrey Madison (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), and Trevor J. Cox (Acoust. Eng., Univ. of Salford, Salford, United Kingdom)

The use of sound diffusors started in antiquity in the form of statuary, balustrades, coffered ceilings, and surface ornamentation. While these surfaces added both beauty and useful scattering, their bandwidth was limited. It was not until the invention of the reflection phase grating by Manfred Schroeder in 1973 that acousticians were able to design number theoretic surfaces with a specified broad bandwidth. Later Cox and D'Antonio expanded diffusive design options beyond number theoretic surfaces to include fractal surfaces, binary amplitude diffusors and beautiful curvilinear architectural shapes acceptable to the architectural community, using shape optimization software. Many peer review publications and books by the authors, ushered in the widespread use of sound diffusion to complement sound absorption. Significant research ensued to study various shapes, bandwidth, location, coverage, and their perceptual aspects. This presentation will review how sound diffusion modified the sound fields originally in recording control rooms, followed by home theaters, rehearsal spaces, performance stages and auditoria, worship spaces, and sound reproduction spaces. Many applications will be presented, as well as subjective questionnaires and objective measures.

1:25

1pAA2. Improving scattering surface design with rapid feedback by integrating parametric models and acoustic simulation. Louena Shtrepi, Jessica Menichelli (DENERG, Politecnico di Torino, C.so DC Degli Abruzzi 24, Torino 10129, Italy, louena.shtrepi@polito.it), Arianna Astolfi (DENERG, Politecnico di Torino, Turin, Italy), Tomas Mendez Echenagucia (Inst. of Technol. in Architecture Block Res. Group, ETH Zurich, Zurich, Switzerland), and Marco C. Masoero (DENERG, Politecnico di Torino, Torino, Italy)

Acoustic panels need to be optimal for both acoustic performance and aesthetic values, in order to respond to the requirements of both architects and acousticians. However, current practice shows that there is a gap in interaction between these professionals. Hence, new strategies in enhancing the efficiency of the design should be investigated. This study proposes a new design process for diffusive surfaces by integrating parametric models and acoustic simulation, aiming to provide architects with rapid visual and acoustic feedback. The new process consists of three parts: (1) basic geometric guidelines for diffusive surface design, which give the necessary information to architects with no acoustic background; (2) programming a parametric model of a diffusive surface sample in *Rhinoceros,* which allows for very quick variations in the design and suits the flexibility required by architects' design practice; (3) connecting *Rhinoceros* and *a simple ray tracing*, which adds acoustic simulation functionality and visual output to facilitate architects' utilization and feedback comprehension. Finally, a characterization of the scattering coefficient is performed in a scale model based on ISO 17497-1 in order to verify the effectiveness of the new design process.

1:45

1pAA3. Exploration of perceptual differences of virtually enhanced sound fields using timbre toolbox. Song Hui Chon and Sungyoung Kim (Elec., Comput., and Telecommunications Eng. Technol., Rochester Inst. of Technol., ECT Eng. Technol., ENT Bldg. 82-2155, 78 Lomb Memorial Dr., Rochester, NY 14623, songhui.chon@rit.edu)

In this presentation, we approach the analysis of perceived spatial differences from the perspective of timbre using the Timbre Toolbox [Peeters *et al*., 2011, *J. Acoust. Soc. Am.*, 130, 2902–2916]. The Timbre Toolbox, using MATLAB, calculates 88 descriptors that have been reported in various timbre perception literature. These include temporal (such as RMS energy), spectral (both in magnitude and power spectra), and harmonic descriptors, as well as those based on the Equivalent Rectangular Bandwidth (ERB) model, which approximates the auditory processing. Analyses were performed on 27 stimuli of nine string quartet groups performing one excerpt in three room conditions, as well as the data from perceptual experiments reported in an earlier study [Chon *et al*., 2015, Proc. Audio Eng.

Soc.]. The three room conditions have different spatial profiles, which consist of one natural room and two virtually enhanced rooms, where many physical parameters were closely controlled (e.g., ST1 and ST2). The experimental data include both the musicians' ratings of spatial quality for performance (e.g., tonal balance and envelopment) as well as recording engineers' ratings of recorded spaces (e.g., clarity and room size). The best timbre descriptor for each question will be reported, and an overall merit of this approach discussed.

2:05

1pAA4. Distinguishing between varying amounts of diffusion subjectively and objectively. Jay Bliefnick and Lily M. Wang (Durham School of Architectural Eng. & Construction, Univ. of Nebraska-Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, jbliefnick@huskers.unl.edu)

This paper summarizes findings from a study of both subjective perception and objective quantification of varying amounts of diffusion. Numerous impulse response measurements were collected from a physical acoustics testing facility that featured reversible absorptive/diffusive/reflective wall panels. The collected room impulse response measurements were utilized in subjective perception trials as well as an objective metric analysis. Subjective testing used auralizations from the measured impulse responses in audio comparison trials to determine how well diffusive room conditions could be discerned. It was found that more than 50% coverage of diffusive surface area was required for the average subject to discriminate between diffusive and absorptive wall conditions. Subjects were even less capable of discerning between diffusive and reflective wall conditions. The objective metric analysis identified the Number of Peaks methodology as the most effective metric for assessing diffusive room conditions from amongst those tested.

2:25

1pAA5. Assessing perceived diffuseness in music venues. Jin Yong Jeon and Kee Hyun Kwak (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, jyjeon@hanyang.ac.kr)

The effect of diffusing elements on the variance of indoor acoustical quality is defined as "diffuseness," and the perception of diffuse sounds is mainly affected by the absorption and diffusion coefficients of surface materials in performance spaces. To characterize sound-field diffuseness, the acoustic parameters of a room, along with impulse-based diffusivity indices are calculated from the binaural room impulse responses. Then, auditory experiments are carried out for using convolved sounds, by investigating the acoustical attributes defined for perceived diffuseness: density of reflection, smoothness of decay, smoothness of reflection, and isotropic directivity. Diffuseness normally reduces the bass ratio and increases the brilliance, thereby increasing the clarity but decreasing the sound strength. In addition, the installed diffusers objectively increase envelopment, intimacy, and diffusion perception. It is seen that the direct sound level supported by the amount of diffuse reflections are related to the perception of diffuseness. With regard to the diffuseness for the acoustical preference in a space, the parameter Np, the number of peaks computed for the measured impulse responses, is effective for evaluating the diffuse sound fields influenced by early scattering reflections.

2:45–3:00 Break

3:00

1pAA6. Comparing the binaural recordings of 22.2- and 2-channel music reproduced in three listening rooms. Sungyoung Kim (RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com), Madhu Ashok (Univ. of Rochester, Rochester, NY), Richard L. King (McGill Univ., Montreal, QC, Canada), and Toru Kamekawa (Tokyo Univ. of the Arts, Tokyo, Japan)

A room interacts with sound sources. It alters both timbral and spatial impression of produced and reproduced sound field(s), and affects the overall sonic experiences. In this study, we approached the room-induced effect in the context of an immersive audio rendering, and investigated a relationship between room acoustics and audio reproduction formats. First, three classical music pieces were recorded and mixed optimally for the 22.2- and 2-channel reproduction formats. Subsequently, we generated a set of binaural responses to each of 22.2- and 2-channel reproduced music at three distinct listening rooms (varying dimensions and reflecting surfaces). Eleven listeners participated in a listening experiment; they compared two randomly selected binaural stimuli and rated perceived dissimilarity. The collected ratings were analyzed through the individual differential scaling (INDSCAL) to determine a perceptual space of the stimuli. The results show that (1) the listeners perceived the stimuli differences through two factors-the reproduction format (dimension 1) and the listening room (dimension 2); and (2) the room-induced perceptual difference was less dominant for the listeners compared to the format-induced one. The importance of each dimension was determined via the goodness of fit of the INDSCAL results as follows: Dimension 1-0.7138 and Dimension 2-0.1649.

3:20

1pAA7. On the effect of acoustic diffusers in comparison to absorbers on the subjectively perceived quality of speech in ordinary rooms. Ali Sanavi, Beat Schäffer, Kurt Heutschi, and Kurt Eggenschwiler (Lab. for Acoust. and Noise Control, Empa - Swiss Federal Labs. for Mater. Sci. and Technol., Überlandstrasse 129, Dübendorf 8600, Switzerland, a.sanavi@gmail.com)

Acoustic diffusers are used to control unwanted reflections responsible for degrading sound quality, or to increase sound diffuseness in spaces such as auditoria. To control unwanted reflections, diffusers are not the only solution. An alternative approach is the use of acoustic absorbers. Whether diffusers or absorbers are chosen as treatment depends on whether the energy conserved by the application of diffusers improves or detracts other aspects of room acoustics including the subjectively perceived qualities. However, not much is known to date about subjectively perceived qualities, e.g., of speech, in ordinary rooms. The aim of this study was twofold. The first aim was to investigate the effect of acoustic diffusers on the subjectively perceived quality of speech in ordinary rooms. The second aim was to determine if and to what extent there are perceptual differences if the diffuser are replaced by acoustic absorbers. Two separate listening tests were performed with stimuli obtained from the convolution of measured binaural impulse responses of a meeting room with excellent speech intelligibility and speech samples recorded in an anechoic chamber. The results of the listening tests confirm that despite the already excellent speech intelligibility and low values of early decay time, speech quality can be further improved by introducing diffusers or acoustic absorbers, with absorbers improving the subjectively perceived speech quality slightly more than the diffusers.

3:40

1pAA8. Theoretical model of reverberation decay in a rectangular room—Potential for predicting flutter echo. Toshiki Hanyu (Junior College, Dept. of Architecture and Living Design, Nihon Univ., 7-24-1, Narashinodai, Funabashi, Chiba 274-8501, Japan, hanyu.toshiki@nihon-u.ac.jp)

Reverberation is the most important factor governing the acoustic design of a room. It is known that non-exponential reverberation decay may occur in a room with unevenly distributed sound absorption, namely, with non-diffuse sound field. We propose the theoretical model of reverberation decay in non-diffuse sound fields. The reverberation decay is characterized by absorption and scattering properties of the surfaces of a room. As a first step for developing reverberation theory of non-diffuse sound field, the proposed model deals with rectangular rooms, because rectangular rooms are very common in any kind of buildings and architecture. Spatial absorption coefficients for three orthogonal directions are defined as the parameters which characterize the energy decay curve in the rectangular room. Within this model, the reverberation in such rectangular room becomes a combination of exponential and power law decay. The power law dependence as a function of time has a singularity. This issue is resolved within the model. The model was verified by comparison with the results of computer simulation using sound ray tracing method. Theoretical solutions of the proposed model agree well with the results of the computer simulation. Potential for predicting flutter echo by this model is also discussed.

4:00

1pAA9. Acoustic design of teachers' cafeteria in Tsinghua University. Xiang Yan (School of architecture, Tsinghua Univ., Rm. 104, Main Academic Bldg., Haidian District, Beijing 100084, China, yx@abcd.edu.cn) and Hui Li (Deshang Acoust., Beijing, China)

By the end of 2015, Tsinghua University has built a new cafeteria for 600 people. Dining period in university campus is not just eating, but an important conversation opportunity. The university president insists the noisy environment in past campus cafeteria must not happen again. So a comprehensive architectural acoustic design was taken. In order to meet the architect's aesthetic requirements, the restaurant uses a large area of seamless porous sandstone sound-absorbing material. By both ceilings/walls absorption treatments and dining tables layout, three acoustic design goals were reached: (1) mid-frequency reverberation time = 0.7s; (2) at the same table, the speaker's sound level attenuation of not more than 6dB, and passed to the adjacent table sound level attenuation of more than 9dB, this ensures the hearing's signal to noise ratio; (3) in busy meal peak, the average background noise does not exceed 65dB (A). Teachers are satisfied with the sound environment of the cafeteria.

MONDAY AFTERNOON, 4 DECEMBER 2017 — SALON F/G/H, 1:00 P.M. TO 4:20 P.M.

Session 1pAB

Animal Bioacoustics and Acoustical Oceanography: Bioacoustic Contributions to the Characteristics and Variability of Soundscapes, Underwater, or Terrestrial

Bruce Martin, Cochair
JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada

David R. Barclay, Cochair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada

Chair's Introduction—1:00

Contributed Papers

1:05

1pAB1. Investigating the performance of soundscape metrics using known data sources and numerical simulations. Bruce Martin (Oceanogr., Dalhousie Univ., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Aidan Cole (JASCO Appl. Sci., Halifax, NS, Canada)

Metrics such as the acoustic complexity index, acoustic diversity index, entropy, evenness, and roughness have been correlated with anthropogenic and biologic sound sources in terrestrial soundscapes. The metrics offer the possibility of characterizing the presence of sound sources and the biodiversity of an environment in large datasets without having to perform detailed automated or manual analysis. However, the metrics are less successful at separating soundscape components and measuring biodiversity when applied to marine soundscapes. The reasons provided for the are the spectral overlap between anthropogenic and biologic sources, the wide range of source levels, long propagation ranges, and the difficulty comparing acoustics to observations, especially for species that are soniferous at night and hidden by day. In this analysis, we examine several popular soundscape metrics for marine data containing typical ambient noise, anthropogenic sources, or a single species. We then examine how the metrics change as different magnitudes and repetition rates of simulated marine life vocalizations are added to the data files.

1:20

1pAB2. Improving the evaluation of soundscape variability via blind source separation. Tzu-Hao Lin (Res. Ctr. for Information Technol. Innovation, Academia Sinica, 128 Academia Rd., Section 2, Nankang, Taipei 115, Taiwan, schonkopf@gmail.com), Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan), Mao-Ning Tuanmu, Chun-Chia Huang (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan), Chiou-Ju Yao (National Museum of Natural Sci., Taichung, Taiwan), Shih-Hua Fang (Dept. of Elec. Eng., Yuan Ze Univ., Chung-Li, Taiwan), and Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan)

Evaluation of soundscape variability is essential for acoustic-based biodiversity monitoring. To study biodiversity change, many researchers tried to quantify the complexity of biological sound. However, the analysis of biological sound remains difficult because the soundscape is made up of multiple sound sources. To facilitate the acoustic analysis, we have applied non-negative matrix factorization (NMF) to separate different sound sources in an unsupervised manner. NMF is a self-learning algorithm which factorizes a non-negative matrix as a basis matrix and an encoding matrix. Based on the periodicity information learned from the encoding matrix, biological chorus and the other noise sources can be efficiently separated. Besides, vocalizations of different species can also be separated by using the encoding information learned from multiple layers of NMF and convolutive NMF. In this presentation, we will demonstrate the application of NMF-based blind source separation in the analysis of long-duration field recordings. Our preliminary results suggest that NMF-based blind source separation can effectively recognize biological and non-biological sounds without any learning database. It can also accurately separate different vocalizing animals and improve acoustic-based biodiversity monitoring in a noisy environment.

1:35

1pAB3. Soundscape components in a shallow-water zone off the coast of Goa. Shyam Kumar Madhusudhana and Bishwajit Chakraborty (Geological Oceanogr. Dept., National Inst. of Oceanogr., CSIR - National Inst. of Oceanogr., Dona Paula, Goa 403004, India, bishwajit@nio.org)

Underwater acoustic data were collected around Grande Island off the coast of Goa, India using moored autonomous recording equipment during March 2016 and March 2017 for four and ten days, respectively. The study site, a shallow-water area in the vicinity of the Zuari river estuary and in proximity of a major shipping port, is presumed to lie along cetacean migration routes. Two different locations were chosen for data collection during each year—a reef site and an off-reef site. All four datasets were dominated by circadian choruses of fish and snapping shrimp. Fish species that were found to contribute significantly to the soundscape includes terapontidae, toadfish, and sciaenidae. Other biological contributions included vocalizations from cetaceans. For example, beaked whale clicks were observed in the 2016 recordings and humpback whale vocalizations were observed for six days in the 2017 recordings. Anthropogenic influence in the soundscape remained low during both years and the only contributing sources were sporadic vessels passing in the vicinity. Given the low measured levels of anthropogenic contribution, the findings of this study could be used as a baseline for performing impact assessments during any future offshore operations in the region.

1:50

1pAB4. Modeling sound fields generated by multi-nodal communications networks found in underwater industrial settings. Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

Increasingly as the sea becomes host to marine industrial operations, mid-to-high frequency multi-nodal communication systems are being employed to monitor equipment states and to control remotely controlled and autonomous underwater vessels used to service and tend underwater equipment. From a regulatory standpoint most of this equipment is modeled as single point noise sources and propagation concerns are ameliorated by high-frequency attenuation due to sound absorption through frequency-dependent chemical "relaxation" or "elasticity" characteristics of boric acid (BH_3O_3) and magnesium sulphate ($MgSO_4$) components in seawater, and assumptions of the directionality of high-frequency signals. But due to the multi-nodal aspect of their operations, these networks actually set up a broad sound field that can be operationally detected at distances in excess of 10km. In this paper we approximate the characteristics of these sound fields by propagation modeling under realistic, but hypothetical scenarios that would be found in offshore and subsea fossil fuel extraction and renewable energy harvesting operations.

2:05

1pAB5. Characterization of diverse coral types using passive acoustic recorded data from the lagoon area of the Kavaratti—An atoll of the Lakshadweep archipelago. Bishwajit Chakraborty (Geological Oceanogr. Dept., National Inst. of Oceanogr., CSIR - National Inst. of Oceanogr., Dona Paula, Goa 403004, India, bishwajit@nio.org), Kranthi Kumar Chanda, and Shyam Kumar Madhusudhana (Geological Oceanogr. Dept., National Inst. of Oceanogr., Goa, India)

The presence of sound in marine habitats can be an indicator of biological processes. Therefore, sound monitoring of the reef system is important to develop an in-depth understanding of the biodiversity, vulnerabilities, and potential changes of ecosystems. In order to understand the diversity related to the coral types, we have computed Acoustic Complexity Index (ACI) (temporal and spectral scales) using passive acoustic recorded data from the lagoon area off the Kavaratti—an atoll of the Lakshadweep archipelago. ACI is found to be the highest in location "A" where the coral type of *Acropora* sp. and ample fish diversity is observed. In location "C" where massive *Porites* sp. and *Pocillopora* sp. with significant acoustic assemblages are observed, ACI was found to be comparatively lower than the location "A." Whereas for "B," "D," and "E" locations, relatively low level of ACI index is obtained. The ACI is the lowest in location "B" where macroalgae with no fish species are seen. Among these three locations, "E" shows higher ACI due to the shrimp sound dominance in the rock substrate having *Porite* sp. coral type.

2:20

1pAB6. Variation in the soundscapes of Pacific coral reefs over multiple spectral, temporal, and spatial scales. Marc Lammers (Hawaii Inst. of Marine Biology, 46-007 Lilipuna Rd., Kaneohe, HI 96744, lammers@hawaii.edu), Eden Zang (Oceanwide Sci. Inst., Makawao, HI), Maxwell B. Kaplan, T Aran Mooney (Woods Hole Oceanographic Inst., Woods Hole, MA), Pollyanna I. Fisher-Pool (National Oceanic and Atmospheric Administration, Haleiwa, HI), and Russell Brainard (National Oceanic and Atmospheric Administration, Honolulu, HI)

Biological sounds occurring on coral reefs are increasingly recognized as important factors influencing reef dynamics and ecological processes. Soundscapes of coral reefs can be broadly divided into a low-frequency band (<1 kHz), dominated by sounds produced by acoustically active fish, and a high-frequency band (2–20 kHz) dominated by snapping shrimp and other invertebrates. Because acoustic activities in both bands are influenced by a variety of ecological (biotic) and environmental (abiotic) factors, coral reef soundscapes are characterized by considerable spatial and temporal variability. The drivers of this variability are not yet well understood, but likely provide important insights into ecosystem processes and condition. We report on an effort to quantify the acoustic activity in both the fish and snapping shrimp frequency bands across twelve coral reef sites in the Pacific Ocean separated by distances ranging from hundreds of meters to thousands of kilometers, including reefs across the Hawaiian Archipelago, the Northern Mariana Islands, and American Samoa. We use data obtained from long-term, bottom-moored acoustic recorders to document the variability observed on multiple temporal scales and examine environmental drivers correlated with this variability at each location and differences among locations.

2:35–2:50 Break

2:50

1pAB7. Contemporary distribution of the eastern North Pacific right whale in the Bering Sea. Dana Wright (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, 3737 Brooklyn Ave., Seattle, WA 98105, dana.wright@noaa.gov), Jessica Crance (Marine Mammal Lab, NOAA Alaska Fisheries Sci. Ctr., Seattle, WA), Daniel Woodrich, Arial Brewer (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, Seattle, WA), and Catherine L. Berchok (Marine Mammal Lab, NOAA Alaska Fisheries Sci. Ctr., Seattle, WA)

The eastern population of the critically endangered North Pacific right whale (NPRW; *Eubalaena japonica*) historically ranged in the eastern Bering Sea from the Aleutian Islands to St. Matthew Island (60.4°N), with limited ($n < 20$) detections further north (which some consider bowhead whale; *Balaena mysticetus*). Since the 1980s, most NPRW sightings have been isolated to the southeastern Bering Sea. In order to describe the current spatio-temporal distribution of NPRW, long-term passive acoustic recorders throughout the Bering Sea (2012–2016) were analyzed manually (10,204.2 days; 27–29% duty cycle) for the presence of NPRW, which were identified using the "up" and "gunshot" calls. NPRW were consistently detected during ice-free months in the southeastern Bering Sea, and intermittently during the same months northward to 59°N. NPRW were also detected at low calling activity within two eastern Aleutian Passes. Notably, NPRW were detected north of St. Matthew (61.6°N) in summer 2016 (July–Aug.). Up and gunshot calls north of 62°N in ice-free months and north of 58°N in ice-associated months could not be distinguished from bowhead whale. Together, these results indicate that NPRW currently range with certainty from the eastern Aleutian Passes to 61.6°N, but may range as far north as the Bering Strait.

3:05

1pAB8. Soundscape fishing: Spatial variability in a low-frequency fish chorus in the southern California kelp forest. Camille M. Pagniello, Jack Butler, Gerald L. D'Spain, Jules Jaffe, Ed Parnell, and Ana Širović (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. #0205, La Jolla, CA 92093-0205, cpagniello@ucsd.edu)

The kelp forests off the coast of southern California support a diverse assemblage of fishes, many of which are known to produce sound. Here, the spatial variability of a low-frequency (325–545 Hz) fish chorus recorded at three sites near the kelp forests off La Jolla, California, is described. This chorus dominated the dusk soundscape at all sites in May/June 2015, 2016, and 2017. During these times, spectral levels around 400 Hz increased by approximately 30 dB over a period of 3 h between 19:00 and 22:00 local time. The location of the fish chorus was estimated during each year using beamforming and time difference of arrival (TDOA) techniques on signals recorded by either a two-element 30-m aperture linear seafloor array or an array with four-elements, 20-m aperture in a tetrahedral-shaped configuration. This location was relatively constant during the chorusing each night. Environmental factors such as temperature, macroalgae assemblage and bottom cover, and geological features were investigated as possible drivers of the spatial distribution of the chorus. [Research supported by California Sea Grant (R/HCME-28) and a Natural Sciences and Engineering Research Council of Canada (NSERC) Postgraduate Scholarship-Doctoral (PGS D-3).]

3:20

1pAB9. Soundscape stability in king penguin colonies. Daniel P. Zitterbart (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 213 Bigelow Lab., MS#11, Woods Hole, MA 02543, dpz@whoi.edu), Camille Toscani (CEFE, CNRS, Montpellier, France), Anna Nesterova, Celine Le Bohec (IPHC, DEPE, CNRS, Strasbourg, France), and Francesco Bonadonna (CEFE, CNRS, Montpellier, France)

King penguin colonies present an acoustically rich environment due to the highly vocal nature of these birds and the large abundance of birds within a colony (100.000 +). Individual identification in adults and chicks is based on vocal signals. Penguins can recognize the call of their partner or chick even among thousands of calling individuals. However, the range of such identification is limited to a few meters, 8.8 m on average in the colony. In spite of the fact that king penguins are famous for their highly developed acoustic abilities, little is known regarding the overall acoustic colony structure. Using a distributed synchronized microphone array, we collected soundscape information over several months at several locations. We find location specific features in the soundscape that are stable over several weeks, regardless of environmental conditions and only require short integration time. We explore if current acoustic indices are able to capture these features, if soundscape stability information might be useful for navigation and if this information can be associated with current colony structure and spatial use.

3:35

1pAB10. An exploration of rhythm perception in African penguins (*Spheniscus demersus*). Irene A. Fobe (Psych., Rochester Inst. of Technol., 1319 Hornblend St., Apt. 8, San Diego, CA 92109, iaf8034@rit.edu), Caroline M. DeLong (Psych., Rochester Inst. of Technol., Rochester, NY), and K. T. Wilcox (Statistics, Rochester Inst. of Technol., Rochester, NY)

Rhythmic properties in penguin vocalizations may be unique to individuals. Rhythm perception is a cognitive ability previously thought to be exclusive to vocal-learning species who have the neurological complexities required to mimic conspecific and heterospecific vocalizations. Discovering rhythm perception in penguins would provide insight on penguins' ability to recognize kin using auditory cues, and discount theories constraining rhythm perception to vocal-learning animals. The goal of this study was to learn if African penguins (*Spheniscus demersus*) could perceive changes in rhythm using a habituation-dishabituation paradigm. Subjects were 32–38 African penguins housed at the Seneca Park Zoo in Rochester, NY. Penguins were played four rhythms at 4 kHz and head turns per bird were counted in 24 sessions. Each session was composed of ten familiarization trials followed by six test trials that alternated between the familiar and novel rhythm. The number of head turns per bird did not significantly increase from the last three familiarization trials to the first novel test trial. Results were inconclusive in showing evidence for auditory rhythm perception in penguins. This may be because subjects met the habituation criterion in only 9 out of 24 sessions. More research on auditory rhythm perception in penguins is needed.

3:50

1pAB11. Aerodynamic Sound Production from Flying Beetles. Rintaro Hayashi, John S. Allen (Mech. Eng., Univ. of Hawaii -Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), and Daniel Jenkins (Molecular BioEng., Univ. of Hawaii Manoa, Honolulu, HI)

The flapping wings of insects generate sounds during their flight. Recent interdisciplinary studies have extensively investigated the lift and thrust mechanisms produced from flapping flight; however, the associated sounds produced are less understood. Most studies have examined the amplitude and frequency with respect to wing beat reporting on the associated harmonic structure. The directionality and phase relationships have been examined in more detail in a few recent studies for flies and mosquitoes. In this study, we examine the sound generation mechanism from two invasive beetle species to Hawaii which are the Oriental Flower Beetle and the Coconut Rhinoceros Beetle. The wing beat, amplitude, and phase relationships are measured and determined with an eight element spherical microphone array. The mechanism of second harmonic generation for the different species is investigated with adaptive signal processing (Empirical Mode Decomposition) and complementary high speed optical video. The rotational motion of wing has a significant role in the second harmonic for the Coconut Rhinoceros Beetle. The different location and phase oscillation of the elytron for the two species results in different vortex generation during the down stroke.

4:05

1pAB12. The soundscape of bat swarms. Laura Kloepper, Yanqing Fu, Morgan Kinniry (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu), Robert L. Stevenson (Univ. of Notre Dame, Notre Dame, IN), Caroline Brighton, Christian Harding (Oxford Univ., Wytham, United Kingdom), Paul Domski (New Mexico Falconry Assoc., Albuquerque, NM), and Graham Taylor (Oxford Univ., Wytham, United Kingdom)

Brazilian free-tailed bats form large maternal colonies numbering in the millions across the American Southwest. Each night, the bats emerge from their roost to travel to foraging locations. During this emergence, individuals fly together in a dense, linear stream and exhibit collective group behavior. Upon morning return to the roost, the flight behavior of bats change and individuals fly in unpredictable paths, with little to no apparent collective group behavior, creating a swarm. To understand the different sensory challenges each of these flight scenarios pose to bats, we recorded the soundscapes of bats in streams and swarms using four different recording platforms: (1) stationary, ground-based directional and omnidirectional microphones, (2) a zip-line microphone that maneuvered through the bat stream and swarm and was monitored by video, (3) a microphone and thermal camera on a quadcopter that recorded bat flight behavior and signals at high altitudes during swarm re-entry, and (4) a trained hawk that flew through the bat stream while carrying a microphone unit and monitored with unique video. We report on the characteristics and variability of soundscapes for bat swarms, including different noise profiles the bats experience during streaming and swarming and the adaptive time-frequency signatures and flight behavior of individuals during group flight.

MONDAY AFTERNOON, 4 DECEMBER 2017 BALCONY M, 1:00 P.M. TO 4:10 P.M.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Acoustic Scattering from Hydrocarbons and Hydrothermal Vents

Daniela Di Iorio, Cochair
Department of Marine Sciences, University of Georgia, 250 Marine Sciences Building, Athens, GA 30602

Alexandra M. Padilla, Cochair
School of Marine Science and Ocean Engineering, University of New Hampshire, Forest Park APT 281, Durham, NH 03824

Christopher Bassett, Cochair
Resource Assessment and Conservation Engineering, National Marine Fisheries Service, Alaska Fisheries Science Center, 7600 Sand Point Way NE, Seattle, WA 98115

Chair's Introduction—1:00

Invited Papers

1:05

1pAO1. Long-term, quantitative observations of seafloor hydrothermal venting using an imaging sonar. Guangyu Xu (Geology and Geophys., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, gxu@whoi.edu), Darrell Jackson (Appl. Phys. Lab. Univ. of Washington, Seattle, WA), Karen G. Bemis (Dept. of Marine and Coastal Sci., Rutgers Univ., NB, NJ), and Anatoliy Ivakin (Appl. Phys. Lab. Univ. of Washington, Seattle, WA)

While connected to the NEPTUNE observatory operated by Ocean Networks Canada (ONC) at the Endeavour Segment of the Juan de Fuca Ridge (http://www.oceannetworks.ca/observatories/pacific), the Cabled Observatory Vent Imaging Sonar (COVIS) recorded an unprecedented long-term (>4 years) acoustic dataset capturing local hydrothermal venting. Processing of the acoustic backscatter data yields three-dimensional (3-D) images of plumes rising tens of meters from black smoker vents on a sulfide structure named Grotto. More importantly, analysis of the Doppler frequency shift in acoustic backscatter yields estimates of the flow rates of those plumes and their volume fluxes. Subsequent calculations based on the vertical variation in volume flux and a theoretical heat-to-volume-flux relationship for buoyancy-driven plumes give estimates of plume heat flux, which are essential for studying the temporal evolution of a hydrothermal system and its coupling with geological, oceanic, and biological processes. In addition to black-smoker plume observations, the ping-to-ping decorrelation of seafloor backscatter recorded by COVIS provides an acoustic indicator of diffuse-flow (i.e., low-temperature, clear hydrothermal discharge) distribution over Grotto and its surrounding areas. Furthermore, acoustic techniques for quantifying the temperature fluctuations, and ultimately, heat flux of diffuse-flow venting are currently under development. [Work supported by NSF.]

1:25

1pAO2. Observing the evolution and fate of free methane in the ocean. Thomas C. Weber, Elizabeth F. Weidner, Alexandra M. Padilla, Kevin M. Rychert, and Scott Loranger (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Free methane gas is increasingly observed escaping the seabed in the world's oceans from sources that are either biogenic, typically in shallow sediments, or from deeper geologic reservoirs. Methane gas bubbles undergo a complicated journey as they rise toward the sea surface. Some or all of the methane may pass through the gas-liquid boundary, into aqueous solution, where it is eventually oxidized and can impact ocean chemistry. Some of the methane, particularly in shallow environments, may reach the atmosphere where it acts as a strong greenhouse gas. One of the key questions regarding the transport of methane upward from the seabed is how much goes where, and this question is being increasingly addressed using acoustic remote sensing techniques. Answering this question begins with seep detection and localization, now routinely performed on data collected with split-beam and multibeam echo sounders. Once located, observations of the bubble-plume backscattering cross section can be used to address questions of flux and vertical gas transport. Both narrow- and broad-band techniques and some of the associated challenges, including wobbly bubbles and multiple scattering in dense plumes, will be discussed.

Contributed Papers

1:45

1pAO3. Investigating bubble transport and fate in the watercolumn with calibrated broadband split-beam echosounder data. Elizabeth F. Weidner (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu), Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH), and Larry Mayer (Earth Sci., Univ. of New Hampshire, Durham, NH)

The transport and eventual fate of gas bubbles in the marine environment is a topic of interest to researchers in numerous fields. Acoustic systems are commonly used to study bubble ebullition sites as they provide synoptic measurements of the watercolumn. However, the visualization of individual bubbles has traditionally required the use of point-source equipment, such as vehicle-mounted cameras. Here, we present an acoustic methodology for studying individual bubbles using a calibrated broadband split-beam echosounder. The extended bandwidth (14–24 kHz) provides a vertical resolution on the order of 10 cm, which allows for the discrimination of individual bubbles in the echogram. Split-beam phase differentiation provides phase-angle data which can be used to compensate for beam-pattern effects and precisely locate bubbles within the watercolumn. Bubble target strength is measured and compared to analytical models to estimate bubble radius, and bubbles are tracked through the watercolumn to estimate rise velocity. The resulting range of bubble radii (1–6 mm in radius) is similar to those found in other investigations, and the rise velocities are consistent with published models. Together, the observations of bubble radius and rise velocity offer a measure of gas flux.

2:00

1pAO4. Evidence of low-frequency multiple scattering of methane gas bubbles at Coal Oil Point, Santa Barbara, California. Alexandra M. Padilla, Scott Loranger, and Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Forest Park APT 281, Durham, NH 03824, apadilla@ccom.unh.edu)

Coal Oil Point, located in the Santa Barbara Channel, CA, is known for its prolific natural hydrocarbon seepage. These hydrocarbons are released at depths between 30 and 80 m, shallow enough that the hydrocarbons reach the surface. Transport via bubbles and droplets through the water column enables the exchange of hydrocarbons between the ocean and the atmosphere. Hydroacoustic observations from previous surveys have been used to map hydrocarbon seep distributions and estimate the flux of hydrocarbons from the seafloor in Coal Oil Point. On September 14, 2016, this area was revisited and surveyed using low-frequency sub-bottom profilers operating from 1 to 10 kHz. The goal is to estimate the gas flux of hydrocarbons from the hydroacoustic measurements and compare them to historic estimates for this area. Visual observations of dense bubble cloud structures of these seeps were seen and evidence of multiple scattering effects were observed in the hydroacoustic measurements. We examine how acoustic scattering of natural hydrocarbon gas bubbles is effected by multiple scattering and how this process affects the quantification of gas flux.

2:15–2:30 Break

2:30

1pAO5. Enhanced subsea leakage detection with broadband active acoustic sensor system. Geir Pedersen (Christian Michelsen Res. AS, P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@cmr.no)

For field re-activation, improved/enhanced oil recovery through injection of gas in reservoirs, and geological storage of CO_2 (carbon dioxide), robust barriers must be provided and confirmed to avoid leakage. Natural seeps from the sea floor represents an additional challenge, as many detection systems will misinterpret them as an infrastructure leakage. Active acoustics (backscattering) has been successfully used for detection of even moderate underwater hydrocarbon gas releases at long ranges, using, e.g., scientific echosounders. However, CO_2 leakage behavior is particularly complex and operations in cold and deep waters make such leakages difficult to detect with active acoustics, due to its behavior and low acoustic impedance in liquid phase. Methodology for detecting CO_2 even in liquid phase is a prerequisite for many deep water operations, and at present no adequate technology exists for this purpose. In order to design an active acoustic system that can detect subsea leakage of hydrocarbons and CO_2 in liquid phase, the behavior and backscattering by droplets as a function of environmental parameters and acoustic frequency is simulated. Theoretical backscattering is further compared with *in situ* acoustic and optical measurements of CO_2, CH_4 (methane), and air, released and measured at depths from 1300m to surface, using a custom built gas release and measurement frame.

2:45

1pAO6. Measurements of methane bubble size distribution from a remotely operated platform in the presence of oil. Theodore F. Argo (Appl. Res. Assoc., Inc., 7921 Shaffer Parkway, Littleton, CO 80127, targo@ara.com) and Paul Panetta (Appl. Res. Assoc., Inc., Gloucester Point, VA)

Subsurface releases of crude oil and methane gas may occur naturally or due to manmade, sometimes catastrophic, events. To determine the quantity of oil and gas being released, measurements can be performed on both the crude oil droplets and methane bubbles to determine their sizes and concentration in the water column. An acoustic backscatter measurement, in the 20–110 kHz range, was developed using transducers mounted on a commercial ROV insonifying an intentional subsurface oil and gas release in a large wave tank. Bubble size distributions for methane mixed in a plume of crude oil were obtained by performing an inversion on acoustic backscattering measurements after accounting for transducer properties and sound transmission through the water column. Due to the rapid nature of the measurements, monitoring of the evolution of the plume in time was also possible.

Distributions were consistent with independent measurements using laser scattering techniques.

3:00

1pAO7. Acoustic investigations of natural seeps at GC600. Mahdi Razaz, Daniela Di Iorio, and James B. Kelly (Marine Sci., Univ. of Georgia, 247, Marine Sci. Bldg., 325 Sanford Dr., Athens, GA 30602, mrazaz@uga.edu)

We examine the acoustic signature of natural seeps in the Green Canyon block 600 of the Gulf of Mexico. The survey site was 2200'1600 m^2 at a depth of 1200 m. Near-bottom multibeam backscatter, side-scan sonar mosaic, and chirp sub-bottom profiles were collected using an AUV cruising at 40 m above the seafloor. The water-column profiles collected by the Kongsberg EM2040 at 200 kHz central frequency were utilized to detect and localize seeps. Geological information from the chirp sonar and side scan images will also augment these data to further our understanding of the geologic structure of natural seeps. Identified seeps from the backscatter intensity were visited with a Comanche ROV for a visual inspection. An oily plume was selected for further investigations and vertical upwelling will be monitored with an Acoustic Scintillation Flow Meter (ASFM), for the first time. Concurrent measurements include nearby Acoustic Doppler profilers, a vertical array of conductivity/temperature instruments and two video cameras (VTLC) for identifying bubble/droplet sizes and distribution and will provide invaluable reference data on temporal variability.

Invited Paper

3:15

1pAO8. Detection and characterization of hydrocarbon droplets using broadband echosounders. Scott Loranger and Thomas C. Weber (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu)

Investigation of the fate and transport of liquid hydrocarbons is limited by the small field of view of current instrumentation. Mass spectrometers, fluorometers, and megahertz sonars—the typical instrumentation for detection and classification of liquid hydrocarbons in the marine environment are limited to detections are ranges of less than a few tens of meters. Lower frequency (80–500 kHz) broadband acoustic backscattering from weakly scattering liquid hydrocarbon targets has been investigated using a novel droplet making device. Results show that such instrumentation should be capable of detections at significantly greater ranges than current instrumentation. The results are compared to a variety of models of acoustic scattering from spherical targets to determine the most accurate model for predicting the frequency response of weakly scattering spheres. The frequency response can be used to characterize the liquid hydrocarbon droplets, as long as the acoustic impedance of the hydrocarbon is well known for the range of temperatures and pressures affecting the droplet.

Contributed Paper

3:35

1pAO9. The acoustic properties of three crude oils at oceanographically relevant temperatures and pressures. Scott Loranger (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu), Justin P. Cole (Chemistry, Univ. of New Hampshire, Durham, NH), Christopher Bassett (Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Seattle, WA), and Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH)

The detection and quantification of crude oil in ocean environments is dependent on adequately constrained acoustic properties (e.g., density and sound speed). However, there is a paucity of published acoustic property measurements of crude oil at oceanographically relevant temperatures and pressures. Three medium crude oil samples (Alaska North Slope, Angola Bavuca, and Angola Xikomba) were tested to better constrain these properties for oceanographic applications. A temperature (-10 to $30\,^{\circ}$C) and pressure (0.1 to 19.3 MPa) controlled sound speed chamber was developed for highly accurate differential time of flight measurements. Density and viscosity were also measured over the same temperature range. Finally, differential scanning calorimetry measurements (-40 to $50\,^{\circ}$C) were conducted to identify phase changes in crude oil constituents that may contribute to nonlinearities in acoustic properties as a function of temperature at a constant pressure. Results are compared to previously available models for sound speed, such as the PC-Shaft model, and density as a function of temperature and pressure. The results can also be used to fully constrain models of the shape of oil droplets in the marine environment as a function of size, an important input for models of acoustic scattering.

3:50–4:10 Panel Discussion

Session 1pEAa

Engineering Acoustics: General Topics in Engineering Acoustics III

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

Contributed Papers

1:00

1pEAa1. A preprocessing method with Hankel matrix-based SVD in strong interference environment. Ge Yu, Shenchun Piao, Xiao Han, and Kashif Iqbal (Acoust. Sci. and Technol. Lab., Harbin Eng. Univ., Harbin, Harbin 150001, China, yuge221@hrbeu.edu.cn)

In sonar system, reverberation is an ignored interference as its characteristic is as same as target echo. It is hard to distinguish target in sever environment. Therefore, suppressing reverberation to increase signal-to-noise ratio is very important in underwater signal processing. In this paper, we proposed a pre-processing method which is based on singular value decomposition and Hankel matrix. This method utilized the correlation between target and reverberation is bad to detect target. The proposed method can suppress both the white noise and reverberation as is evident by the simulation results. The proposed method is also verified by processing the experimental data.

1:15

1pEAa2. Rotor noise control using 3D-printed porous materials. Con Doolan, Chaoyang Jiang (School of Mech. and Manufacturing Eng., Univ. of New South Wales, UNSW Sydney, Sydney, NSW 2052, Australia, c.doolan@unsw.edu.au), Danielle Moreau (Mech. and Manufacturing Eng., Univ. of New South Wales, Adelaide, South Australia, Australia), and Yendrew Yauwenas (Mech. and Manufacturing Eng., Univ. of New South Wales, Sydney, NSW, Australia)

The self-noise created by rotor blades is an important component of the noise generated by aeroengine fans, wind turbines, and propellers. It is created by the passage of boundary layer turbulence over the trailing edge of the blade, where a portion of the unsteady pressure near-field is converted to acoustic waves that subsequently propagate to an observer in the far-field. This paper explores the use of 3D-printed porous materials for controlling rotor self-noise. It is reasoned that the flexibility offered by 3D-printing makes it possible to produce blade trailing edges with tailored, spatially varying porosity that can maximise noise reduction, either by modifying the near-field turbulence or by affecting the production of waves at the trailing edge itself. A number of 3D-printed porous material samples were acoustically characterised using an impedance tube. Using this information as a guide, special rotor blade tips were 3D-printed with varying levels of porosity and evaluated using a rotor-rig and microphone array. Simultaneously, computational modelling of the rotor-rig was performed to understand the flow field and turbulence levels over the blade tip region.

1:30

1pEAa3. On the character and mechanics of flow-induced wall-mounted finite airfoil noise production. Danielle Moreau and Con Doolan (School of Mech. and Manufacturing Eng., UNSW, UNSW, Sydney, NSW 2052, Australia, d.moreau@unsw.edu.au)

A major source of noise involves flow interaction with wall-mounted attachments such as fins, guide vanes, aircraft airframe components, and wind turbine blades. In a simplified sense, these objects can be viewed as wall-mounted airfoils of finite length. A wall-mounted finite airfoil has one end immersed in the free stream and the other end fixed to a wall that is subject to a developing boundary layer. The free-end and wall junction introduce three-dimensional flow structures into the wake that affect noise generation in ways that are not yet fully understood. The primary goal of this research is to determine how airfoil end-effects and three-dimensional aerodynamics influence airfoil noise production. To accomplish this goal, anechoic wind tunnel studies have been conducted to determine the characteristics and mechanisms of flow-induced noise production for various wall-mounted finite airfoil configurations. The test models include combinations of airfoil profile shape, aspect ratio, angle of attack and free stream velocity. The motivation for this work is to gain a better understanding of sound generation from wall-mounted finite airfoils to aid future innovation in quiet aerodynamic design and optimisation.

1:45

1pEAa4. An experimental study on the role and function of the diaphragm in modern acoustic stethoscopes. Lukasz Nowak (Inst. of Fundamental Technolog. Res., Polish Acad. of Sci., ul. Pawinskiego 5B, Warszawa 02-106, Poland, lnowak@ippt.pan.pl)

Vibrations of a diaphragm of an acoustic stethoscope in contact with the body of an auscultated patient are the source of the sound transmitted to the ears of a physician performing examination. Mechanical properties of a diaphragm are supposed to significantly affect the parameters of the transmitted bioacoustic signals. However, the exact relation remains mostly unclear, as the underlying phenomena involve complex effects of acoustic coupling between the diaphragm and the body of a patient. The present study introduces a detailed methodology for determining vibroacoustic behavior of a diaphragm of a stethoscope during an auscultation examination. A laser Doppler vibrometer is used to measure the velocity of various points on the surface of a diaphragm during heart auscultation. Synchronized recordings of electrocardiography signals are used for segmentation. Representative data sets are selected and analyzed for various kinds of diaphragms. The results show significant differences in vibration velocity levels and their distribution across the surfaces of the considered structures. In this regard, it is also shown that the currently available solutions can be significantly improved by using structures better matched to the acoustic impedance of a body.

2:00

1pEAa5. A study on a friendly automobile klaxon production with rhythm. SangHwi Jee, Myungsook Kim, and Myungjin Bae (Sori Sound Enginnering Lab, Soongsil University, Seoul 06978, South Korea, slayernights@ssu.ac.kr)

We are always exposed to traffic noise in our lives. Especially in unproven situations, exposure to a sudden sound may result in loss of auditory cells and exposure to retaliatory driving. However, if you do not use horn sounds, you will not be free from traffic accidents when driving. The Clackson is used as a car horn and has a simple mechanical structure, which is easy to use. However, it has a disadvantage that it can not redesign the sound pressure of fluid sound. In this study, the power supply time width of Clackson was adjusted to five (0.01 s, 0.02 s, 0.03s, 0.06 s, and 0.13 s). The

result was a sound level of five (80 dB, 85 dB, 90 dB, 100 dB, and 110 dB) sound levels of the Klaxon. The experimental result shows that the maximum sound pressure (pmax = 110 dB) after operating the clackson is tmax. Ps (dB) = 110 (dB)—{10log (ton / (ton + toff)) + 20log (ton / tmax) The preference was evaluated for five types of Clackson sounds of 5 seconds. The MOS evaluation was performed on 100 participants after three times of five kinds of Klaxon sounds. The evaluation items were MOS measurement for risk, sound size, unpleasantness, and stress. As a result of evaluation, when designing various forms of clackson sound, it was quickly felt that it is danger to give a sound rhythm to a horn sound rather than a conventional sound. In particular, if you change the rhythm of the sound of Klaxon, the risk perception becomes faster even though the average sound level is lowered by −20 dB.

MONDAY AFTERNOON, 4 DECEMBER 2017 — BALCONY N, 2:45 P.M. TO 4:15 P.M.

Session 1pEAb

Engineering Acoustics: General Topics Engineering Acoustics IV

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

Contributed Papers

2:45

1pEAb1. Comparison and analysis of the methods defined by ASTM standard E2235-04, ISO 3382-2-2008, and EASE acoustical modeling software to determine reverberation time RT60 in ordinary rooms. Juan C. Montoya (none, 41 Churchill St., Springfield, MA 01108, jmontoya@csacoustics.com)

Determining the reverberation time RT60 is a well known acoustical parameter used by acousticians to characterize acoustically a room. This paper discusses the differences in the results obtained from measuring RT60 using common standards methods defined by ASTM and ISO. It also compares and contrasts those results with acoustical modeling software (EASE) for a small office room, a gymnasium, and a small church sanctuary. A discussion between the procedures utilized, which includes type and location of sound source, position and requirements of the measurement microphones, repeatability of the measurement sample, signal used for the measurement, and frequency range, are attempted in order to evaluate common ground and a relative best practice.

3:00

1pEAb2. Measurement of speed of sound profile as a function of altitude. Zhuang Li, Brett Schaefer, Brian Schaefer, William Dever, Tyler Morgan, and Matthew Foltz (Chemical, Civil, and Mech. Eng., McNeese State Univ., Box 91735, Lake Charles, LA 70609, zli@mcneese.edu)

The goal of this undergraduate research project is to design a balloon payload of less than 500 g to measure speed of sound and atmospheric data including temperature, pressure, humidity from 0 to 100,000 feet altitude. The payload is made of polystyrene foam due to its lightweight and good thermal isolation property. An ultrasonic sensor with a reflection mirror are installed outside payload box to measure speed of sound. As the ultrasonic sensor's working temperature is above −40 °C, a heater is attached to the sensor but isolated from the environment. Temperature, pressure, and humidity sensors were calibrated. Power consumption was also calculated and power is supplied by 2CR5 batteries. Software for the project was developed in Arduino DUE. An SD card shield is connected to the Arduino board for data storage and RTC clock. Tests were conducted in vacuum chamber and freezer prior to flight. Balloon launch was successfully conducted in the NASA CSBF on May 16 2017. Speed of sound was also calculated using measured environmental data. Measured and calculated speed of sound profiles were compared and reasonably agree with each other. An improved payload will launch in Carbondal, IL, on August 21, 2017, during the solar eclipse.

3:15

1pEAb3. Experimental study of nonlinear acoustic effects of orifices in duct systems. Samuel T. Kawell, Thomas Teasley, and David Scarborough (Aerosp. Eng., Auburn Univ., 211 Eng. Dr., Auburn, AL 36849, stk0011@auburn.edu)

Researchers focused on reduced emissions hydrocarbon fuel combustion systems have developed new high-efficiency furnace and gas turbine engine technologies that, unfortunately, regularly suffer from detrimental combustion instabilities. Engineering design tools developed to predict these instabilities in combustion systems frequently neglect nonlinear acoustic effects. However, boundaries and duct junctions, e.g., area changes, valves, and orifices, often exhibit nonlinear effects even at low acoustic pressure amplitudes. Experimental data of these nonlinear effects are required to accurately model engine and duct acoustics and to predict combustion instabilities. This experimental investigation focuses on understanding the nonlinear acoustic response of an orifice with and without steady flow. Various orifices were mounted in a multiple-microphone impedance tube and the acoustic impedance of the combination was used to measure the nonlinear acoustic response and impedance of the orifices at amplitudes from 114 to 190 dB and over frequencies from 100 to 1500 Hz. The preliminary results of the experimental study suggest that the impedance is dependent on the acoustic velocity amplitude. This indicates significant nonlinear effects. These experimental data will assist the development of nonlinear acoustic models of orifices in combustion systems.

3:30

1pEAb4. Extension of the phase and amplitude gradient estimator method for acoustic intensity to multiple tones. Kelli Succo, Scott D. Sommerfeldt, Kent L. Gee, and Tracianne B. Neilsen (Phys., Brigham Young Univ., N203 ESC, BYU, Provo, UT 84602, kelli.fredrickson7@gmail.com)

The phase and amplitude gradient estimator (PAGE) method [Thomas *et al.*, *J. Acoust. Soc. Am.* **137**, 3366–3376 (2015)] has proven successful in improving the accuracy of measured energy quantities over the traditional p-p method in several applications. One advantage of the PAGE method is the use of phase unwrapping, which allows for increased measurement bandwidth. However, phase unwrapping works best for broadband sources and fields with high coherence. Narrowband sources often do not have coherent phase information over a sufficient bandwidth for a phase unwrapping algorithm to unwrap properly. Even for narrowband signals, the PAGE method has been shown to provide correct intensity measurements for frequencies up to the spatial Nyquist frequency. This is improved bandwidth over the p-p method. Previous work with sawtooth waves in a plane wave tube shows that in cases with multiple tones, the PAGE method with a few additional steps of processing accurately calculates intensity above the spatial Nyquist frequency provided one tone is below it. A variety of further experiments with multiple tones are explored to determine if any extra steps in processing or ingenuity in data acquisition can reasonably be used to achieve comparable results in a free-field environment. [Work supported by NSF.]

3:45

1pEAb5. A new lumped parameter model for the design of the free-flooded ring transducer. Kyounghun Been, Seungwon Nam (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO 416, Hyoja-dong, Nam-gu, Pohang-si, Gyeongbuk 790-784, South Korea, khbeen@postech.ac.kr), Haksue Lee, Hee-seon Seo (Agency for Defense Development, Changwon, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

The free-flooded ring (FFR) transducer is the well-known low-frequency sound sources in underwater because its operating frequency bandwidth is broad and relatively small size. Previous researches were preformed to predict the characteristics of FFR transducers using ECM which is a type of LPM because ECM is widely used to understand its characteristics in transducer design process. However, it is hardly to predict the characteristics of an FFR transducer because the acoustic field is generated from its top and bottom openings, connected by the inner fluid, as well as the cylindrical ring surface. Here, the authors investigated an ECM of an FFR transducer consisting of three parts: the piezoelectric ring, the cylindrical cavity, and the radiation load. In addition, an LPM which can consider mutual radiation loads was proposed to improve the accuracy of the model. The proposed models were verified using commercial finite element method (COMSOL Multiphysics). It was confirmed that LPM could predict characteristics of FFR transducer more accurately than ECM. [Work supported by NRF 2016R1E1A2A02945515).]

4:00

1pEAb6. Fluid instabilities in turbocharger compression systems. 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)

Turbochargers increase the power density of internal combustion engines, which allows the displaced volume to be reduced and the fuel efficiency to be improved for the same level of performance. However, the demand for turbocharger centrifugal compressors to deliver increasingly elevated boost pressures, over a wide air flow range, results in unfavorable flow fields that generate narrow and broadband noises within the engine air induction system. The present study is a combined computational and experimental effort, focusing on identifying unsteady fluid-flow instabilities in a turbocharger compression system installed on a bench top stand. This work concentrates primarily on prediction of mild and deep surge instabilities, since they limit the low-flow operating range. Surge occurs near the Helmholtz resonance frequency of the compression system (including a compressor inlet duct, centrifugal compressor, compressor outlet duct, plenum, plenum outlet duct, and a valve) and results in low frequency pressure fluctuations throughout the compression system. Additionally, the compressor flow field adapts to operation at reduced flow rates by forming rotating instabilities in the impeller inducer and vaneless diffuser, which contribute to the occurrence of surge and produce noise as they interact with the rotating impeller and stationary volute.

MONDAY AFTERNOON, 4 DECEMBER 2017 — STUDIO 4, 1:00 P.M. TO 4:00 P.M.

Session 1pMU

Musical Acoustics: Marching Band Instruments

D. Murray Campbell, Cochair
School of Physics and Astronomy, University of Edinburgh, James Clerk Maxwell Building, Mayfield Road, Edinburgh EH9 3JZ, United Kingdom

Thomas Moore, Cochair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—1:00

Invited Papers

1:05

1pMU1. Saxophone acoustics and marching bands. Vincent Gibiat (ICA, Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr), Jerome Selmer, and Jonathan Cottier (Henri Selmer Paris, Paris, France)

Historically saxophones appear in bands far after the beginning of Jazz history. Marching bands were mainly built around brass instruments and the saxophone appears alone in these bands. The main question that arises is "how this brass-woodwind succeeded to mix in such brass bands? The power of brass instruments is known to cover most woodwinds but the saxophone survives when flutes, with the exception of piccolo, disappeared. The answer could be in the design of saxophone and mainly in the design of tenor saxophone.

The saxophone has been built and designed by A. Sax to be a powerful instrument able to cover or to compete with the sound of brass instruments, and he has been successful. We will show that in addition with the native sound power of its side, the saxophone exhibits a formant in the most sensitive part of our hearing system that gives it such "voicy" sound. Finally, we will show that saxophones are as wild instruments as the other members of marching bands with theses two characteristics power and voice.

1:25

1pMU2. Single-reed woodwinds: From physical modeling to sound radiation. Vasileios Chatziioannou and Alex Hofmann (Dept. of Music Acoust., Univ. of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, chatziioannou@mdw.ac.at)

Single-reed woodwind instruments have been traditionally used in marching bands, along with brass and percussion instruments. However their sound is often covered by louder instruments, making them less audible to the audience and the members of the band. Indeed, the saxophone has been designed in order to replace the clarinet in military bands, making the woodwind section of the band more prominent. This study discusses the sound generation mechanism of single-reed woodwind instruments in an attempt to investigate which reed and mouthpiece properties significantly affect the amplitude of the radiated sound. To this end, transfer function measurements and numerical investigations using physical modeling are employed. The former may indicate how the use of different mouthpiece geometries and various types of reeds may have an effect on the sound magnitude. The latter can be used to systematically vary reed and embouchure related parameters while observing how changes in these parameters affect the radiated sound.

1:45

1pMU3. Traditional lip-blown aerophones in China. Stewart Carter (Music, Wake Forest Univ., 1833 Faculty Dr., Winston-Salem, NC 27106, carter@wfu.edu)

Bands and orchestras in present-day China universally employ Western-style valved trumpets and horns, but "natural" lip-blown instruments were known in China as early as the Han Dynasty, when *guchui* (wind and drum) ensembles played an important role in court rituals and military processions. Drawing on early artworks, treatises, and modern ethnographic studies, my paper demonstrates the enduring use of traditional lip-blown instruments in China, from early imperial times to the present day. The earliest trumpets in China probably were made from animal horns. In the Mogao Grottoes near Dunhuang, on the ancient Silk Road, a mural shows musicians playing animal horns associated with drums, celebrating the defeat of the Tibetan army by Chinese forces in 848, but the trumpets in two brick reliefs dating from approximately three centuries earlier appear to have been made of metal, probably bronze. In China, today natural lip-blown instruments endure in Buddhist ritual music, primarily in Tibet and Mongolia, as well as in certain minority cultures, where they are employed primarily in processions accompanying weddings and funerals.

2:05

1pMU4. Serpents on parade. D. Murray Campbell and Arnold Myers (Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

The serpent which was used to accompany plainchant in French churches from the beginning of the seventeenth century was a wooden lip-excited wind instrument with a wide conical bore and six fingerholes. Toward the end of the eighteenth century, the serpent found another role as a bass instrument in civilian and military bands. The traditional form of the church serpent was modified in various different ways to make it more convenient for playing on the march. This paper compares the acoustical and musical properties of the church serpent, the horizontal English military serpent, and several types of upright serpent designed for band use.

2:25–2:45 Break

2:45

1pMU5. Timbre of marching-band brass instruments. Robert W. Pyle (P & S Horns, 11 Holworthy Pl., Cambridge, MA 02138, rpyle@icloud.com), Sabine Klaus (National Music Museum, Vermillion, SD), and Arnold Myers (Univ. of Edinburgh, Edinburgh, United Kingdom)

The tone quality of brass instruments for outdoor use has varied from time to time and from place to place. Instruments currently in use can be recorded and analyzed. However, many older instruments, typically those in museum collections, are not allowed to be played. Hence, it is necessary to rely on measurements of physical dimensions and acoustic input impedance to form comparisons. In this paper, such measurements are used to estimate two important timbral parameters: the spectral centroid at moderate playing levels and the brassiness potential. For different instruments of the same pitch, like trumpet and flugelhorn, the spectral centroid gives a measure of the brightness or darkness of the tone. The brassiness potential gives a measure of the degree to which the tone develops a "brassy" edge at high playing levels. As one might expect, the trumpet has both a higher spectral centroid and a higher brassiness potential than the flugelhorn. Results will be shown comparing instruments built in different countries and at different times.

3:05

1pMU6. Acoustics, performance, and instrument invention at Cyclophonica Bicycle Orchestra. Leonardo Fuks (Musicology and Music Education, UFRJ Rio de Janeiro Federal Univ., Rua do Passeio 98, Rio de Janeiro, Rio de Janeiro 20021-290, Brazil, fuks.leonardo@gmail.com)

Cyclophonica Bicycle Orchestra was created in 1999 in Rio de Janeiro- Brazil by the present author as a platform for experimenting new ways of music making, developing new instruments and techniques. Inspired by the traditional marching and animal-mounted bands, present all over the world, the project of Cyclophonica aimed at exploring open-air or closed ambients, involving or moving with the audience and being able to control "chamber" music performance and instruments, while ensuring safety to the players and listeners.

Several questions were still open: which were the instruments that could be played?; would musicians be able to accelerate the tempo and reduce the bicycle speed independently?; how well would the performers coordinate music without a conductor or looking to each other?; how would the listeners receive such music that would be produced and diffused in a different way? A group of six musicians was formed in the beginning. Presently, after more than eighteen years later, the Cyclophonica has ten fixed members and two additional ones, and has performed more than two hundred and fifty times, in different festivals, events, and official ceremonies, almost always being paid for, being perhaps the only professional group of this kind in the world.

3:25

1pMU7. A history of brass bugles in American drum and bugle corps. Jack Dostal (Phys. Dept., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

Bugles have long played an important role in American bands and corps, reaching beyond their military origins. As early as the 1920s, competitive drum and bugle corps performed in national competitions sponsored by the American Legion and the Veterans of Foreign Wars. Modern drum corps continue to compete through organizations such as Drum Corps International and Drum Corps Associates. While brass bugles in these competitive drum corps began as military signaling devices, successive modifications made them capable of greater ranges of music. These traditionally valveless, key-of-G bugles evolved to include pistons and rotors, gaining notes beyond a single harmonic series. Instrument sizes were varied to permit pitch ranges from contrabass to soprano. The corresponding music in these shows thus expanded from bugle calls to classical, jazz, and popular genres. This talk will trace the evolution of bugles and their capabilities in American drum corps from their 20th century history through the current day.

Contributed Paper

3:45

1pMU8. Experimental and computational investigation of the acoustics of ErgoSonic Percussion angled shell drums. Mahdi Farahikia, Ronald Miles (Mech. Eng., SUNY Binghamton, 13 Andrea Dr. Apt. A, Vestal, NY 13850, mfarahi1@binghamton.edu), and Ken Turner (ErgoSonic Percussion, Apalachin, NY)

Acoustic principles of ErgoSonic Percussion's patented angled shell drums were studied through the Finite Element Method (FEM) predictions and acoustic measurements in our anechoic chamber. Unlike existing drums, this drum design provides improved playability and the ability to easily modify the resonant decay and character of the sound. When used in marching bands, this design is found to significantly reduce strain and fatigue of the musician. Computational and experimental results are presented to examine the effects of the shape of the drum, and the size, location, and orientation of a port on the resonant head on the pitch, sound decay and overall tuning of the instruments. This angled drum design is found to provide a smaller, lighter, and more ergonomic instrument that produces sound that is essentially equivalent to that of conventional drums. It is also shown that the acoustic properties of the port can be easily adjusted during use to significantly modify the resonant character of the sound by changing the system damping. Adjustment of the decay in the sound of the drums can be easily accomplished by modifying the cover on the port using different porous materials, each of which leads to a different decay rate.

Session 1pNS

Noise and Physical Acoustics: Supersonic Jet and Rocket Noise II

Seiji Tsutsumi, Cochair
JEDI center, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara 252-5210, Japan

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Observations regarding the noise radiated from full-scale heated, supersonic jets. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

In this presentation, the characteristics of supersonic jet noise are reviewed. Observations from more than a decade's worth of measurements of high-performance jet engines and large solid rocket motors are compared against each other and laboratory-scale findings. These include apparent source location and extent, directivity, spectral shape, relative importance of different noise components, and presence of nonlinear propagation effects.

1:25

1pNS2. Spatiotemporal analysis of high-performance military aircraft noise during ground run-up. S. Hales Swift, Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N221 ESC, Provo, UT 84602, hales.swift@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Wright-Patterson Air Force Base, Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Recent measurements of high-performance military aircraft noise have revealed that full-scale jet noise has features and structures that are still only partly understood, such as the presence of multiple acoustic radiation lobes in the aft direction at certain frequencies. Spatiotemporal analyses of a ground-based microphone array measurement of the noise from a tethered F-35 at various engine conditions are used to investigate these features of the sound field. The ground array covered an angular aperture of 35–152 degrees relative to the front of the aircraft. The large angular aperture allows for a detailed investigation of the correlation and coherence at frequencies exhibiting multi-lobe behavior. This spatiotemporal analysis yields further evidence of the characteristics of multi-lobe behavior in high-performance, full-scale jet noise. [Work supported by an Office of Naval Research grant, a USAFRL SBIR, and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 07/10/2017; JSF17-714.]

1:45

1pNS3. Noise sources in a commercial supersonic jet. Christopher J. Ruscher and Sivaram Gogineni (Spectral Enegies, LLC, 5100 Springfield St., Ste. 301, Dayton, OH 45431, cjrusche@gmail.com)

Stringent noise regulations currently limit commercial aviation. These regulations make supersonic commercial flight impractical. The development of an engine that can meet these strict rules is paramount to making supersonic commercial flight a reality. One method of noise reduction is to add additional streams to an engine. As such, the three-stream jet has potential to help reduce exhaust noise. Understanding the noise sources in the jet plume can help to design nozzles that are quieter. To accomplish this, high-fidelity, high-speed data are required. Data for an axisymmetric and offset three-stream nozzle were generated using the LES code JENRE developed by the Naval Research Laboratory. The simulation data has been shown to match well with experimental data. Advanced analyses methods that are based on Proper Orthogonal Decomposition (POD), wavelet decomposition, and Stochastic estimation have been applied to extract noise sources in the jet plume.

2:05

1pNS4. High-performance aircraft short-takeoff and vertical-landing noise measurements on an aircraft carrier. Alan T. Wall, Richard L. McKinley (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Allan C. Aubert, Russell W. Powers, Michael J. Smith, Charles J. Stouffer (Naval Air Systems Command, Naval Air Station Patuxent River, MD), and James C. Ku (Naval Air Systems Command, Navail Air Station Patuxent River, MD)

The noise levels caused by high-performance aircraft are relatively high in the close proximity experienced by crew on board aircraft carriers, which can interfere with communications and may pose a risk for hearing loss. This paper reports on preliminary results of noise measurements of the operations of F-35B aircraft performing short-takeoff and vertical-landing (STOVL) operations on the flight deck of an LHA aircraft carrier. This noise measurement campaign was performed in late 2016, by scientists from the Air Force Research Laboratory (AFRL) in collaboration with the Naval Air Systems Command (NAVAIR) and the F-35 Integrated Task Force (ITF). The measurements were taken using hand-held noise recorder systems, and the recording engineers shadowed actual locations of crew. These data will be used to validate STOVL models of crew noise exposures on deck. [Work supported by F-35 JPO.]

2:25

1pNS5. Characterization of broadband shock-associated noise from high-performance military aircraft. Tracianne B. Neilsen, Aaron Vaughn, Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

For nonideally expanded jets, broadband shock-associated noise (BBSAN) is a feature in the sideline and forward directions. While BBSAN has been studied fairly extensively for laboratory-scale jets, its presence and characteristics in full-scale, tactical aircraft noise need to be evaluated. Noise measurements on a tied-down F-35 provide the opportunity to characterize full-scale BBSAN using a linear ground array that spanned a large angular aperture: 35–152 degrees relative to the front of the aircraft. The main questions are whether the full-scale BBSAN shares the same characteristics as those observed in laboratory-scale BBSAN and if current models capture the features of full-scale BBSAN. The variation in the spectral shape, peak frequency, and peak level of full-scale BBSAN across angle for different engine powers is explored and compared to prior laboratory studies. Comparisons are also made with models for BBSAN based on stochastic theory ([Tam *et al.*, *J. Sound Vib.* **140**, 55–71 (1990)]) and the simplified model used in Kuo *et al.* [AIAA Paper 2011–1032 (2011)] for lab-scale BBSAN. Frequency-dependent convective speed estimates obtained from the current BBSAN models are compared to estimates based on directivity. [Work supported by the Office of Naval Research and the F-35 JPO.]

2:45–3:00 Break

Contributed Papers

3:00

1pNS6. Spectral decomposition of turbulent mixing and broadband shock-associated noise from a high-performance military aircraft. Aaron Vaughn, Tracianne B. Neilsen, Kent L. Gee (Brigham Young Univ., C110 ESC, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Sound from high-performance military aircraft originates primarily from the turbulent mixing noise, but at smaller inlet angles, broadband shock-associated noise (BBSAN) is present. The similarity spectra of the two components of turbulent mixing noise developed by Tam *et al.* [AIAA Paper 96–1716 (1996)] represent noise associated with fine and large-scale turbulent structures and provide reasonable fits for ideally expanded, supersonic jet noise. For non-ideally expanded jet flow, BBSAN contributions to the spectral shape need to be included in spectral decompositions in the sideline and forward directions. A model proposed by Tam *et al.* [*J. Sound Vib.* **140**, 55–71 (1990)] and later simplified by Kuo *et al.* [AIAA Paper 2011–1032 (2011)] provides a spectral function that models the BBSAN spectral shape. The ability of the BBSAN and similarity spectra shapes to account for the measured spectra is evaluated for ground-based microphones that covered a spatial aperture from 35 to 152 degrees. Spectral decompositions at low and high engine powers are compared. Using turbulent mixing noise similarity spectra decomposition in conjunction with BBSAN empirical fits, a better equivalent source model can be developed. [Work supported by the Office of Naval Research and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited.]

3:15

1pNS7. Modeling shock formation and propagation in high-performance jet aircraft noise. Brent O. Reichman, Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Nonlinear propagation can play an important role in both time and frequency-domain features of far-field supersonic jet noise. Many aspects of nonlinear propagation, such as waveform steepening and greater-than-expected high-frequency spectral levels, have been previously predicted for select angles and engine conditions. This paper builds on previous successes and presents a comparison of nonlinear and linear predictions for the F-35B aircraft. Results are shown over a wide spatial and angular range and over varying engine power conditions, including showing evidence of nonlinear propagation in the forward direction at the highest engine conditions. In addition, specific features, such as individual shocks, are compared between numerically propagated and measured waveforms, highlighting the successes and deficiencies of current propagation models. Weather and multipath interference effects are also addressed and corrected using an empirical model. [Work supported by USAFRL through ORISE. Distribution A: Approved for public release; distribution unlimited; Cleared 07/10/2017; JSF17-714.]

3:30

1pNS8. Measurement methods for high-performance jet aircraft noise inside a hardened aircraft shelter. Richard L. McKinley, Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com), Theo A. van Veen (NLR: Nederlands Lucht- en Ruimtevaartcentrum – Netherlands Aerosp. Ctr., The Hague, Netherlands), and Jaap van't Hof (TNO: Nederlandse organisatie voor Toegepast Natuurwetenschappelijk Onderzoek – The Netherlands Organisation for Appl. Sci. Res., The Hague, Netherlands)

High-performance military aircraft regularly operate inside hardened aircraft shelters (HAS). The F-35A aircraft must be certified as safe to operate inside a HAS before it can be deployed and used in such structures worldwide. Acoustic levels at maintainer locations allow for noise dose estimates and regulation of personnel mission support in HASs to mitigate risks of hearing damage. Acoustic levels impinging on the airframe are compared against engineering design limits in order to prevent a reduction in operational lifespan of the aircraft due to acoustic fatigue. The Air Force Research Laboratory, the Royal Netherlands Air Force, and the Dutch national laboratories NLR and TNO collaborated on a set of acoustic measurements for an F-35A operating inside a HAS at Leeuwarden airbase in the Netherlands. The methods, analysis, and qualitative findings of the acoustic measurements are presented here. [Work supported by RNLAF and by the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 01/24/2017; JSF17-035.]

3:45

1pNS9. Calculating the frequency-dependent apparent source location using peak cross-correlation between near-field and far-field microphone arrays. Jacob A. Ward, S. Hales Swift, Kent L. Gee, Tracianne B. Neilsen (Phys., Brigham Young Univ., N243 ESC, Provo, UT 84602, jacob.ward@live.com), Koji Okamoto, and Masahito Akamine (Dept. of Adv. Energy, Graduate School of Frontier Sci., The Univ. of Tokyo, Kashiwa, Chiba, Japan)

The apparent acoustic source region of jet noise varies as a function of frequency. In this study, the variation of the apparent maximum source location with frequency is considered for an ideally expanded, unheated, Mach-1.8 jet with exit diameter of 20 mm and a Reynolds number of 6.58e6. In this study, the source location is ascertained for one-third octave bands by evaluating peak cross-correlation between near-field linear microphone arrays at three sideline distances and a far-field microphone arc. The impact of the hydrodynamic field on correlation results is considered. Source locations determined by these means are compared with intensity analyses for the same jet [K. L. Gee *et al.*, AIAA Paper 2017–3519 (2017)]. Correlational methods together with filtering can provide a straightforward measure of the acoustic origin as a function of frequency and thus inform optimal microphone array layout for specific frequency regimes.

4:00

1pNS10. Numerical validation of using multisource statistically optimized near-field acoustical holography in the vicinity of a high performance military aircraft. Kevin M. Leete (Brigham Young Univ., Provo, UT 84604, kevinmatthewleete@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., Provo, UT), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Multisource statistically optimized nearfield acoustical holography (M-SONAH) is an advanced holography technique [Wall *et al.*, *J. Acoust. Soc. Am.* **137**, 963–975 (2015)] that has been used to reconstruct the acoustic field from measurements taken in the vicinity of a high-performance military aircraft [Wall *et al.* **139**, 1938 (2016)]. The implementation of M-SONAH for tactical jet noise relies on creating an equivalent wave model using two cylindrical sources, one along the jet centerline and one below the ground as an image source, to represent the field surrounding an aircraft tethered to a reflecting ground run up pad. In this study, the spatial and frequency limitations of using the M-SONAH method to describe the field of a tethered F-35 is explored by using the same measurement geometry as at a recent test, but substituting the sound field obtained from a numerical source for the measurement data. The M-SONAH reconstructions are then compared to numerical benchmarks. A spatial region and frequency bandwidth where bias errors are low are identified and provide validation for the use of this method in tactical jet noise source and field reconstructions. [Work supported by USAFRL through ORISE and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 07/10/2017; JSF17-714.]

4:15

1pNS11. Sound quality analysis of far-field noise from a high-performance military aircraft. S. Hales Swift, Kent L. Gee, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N221 ESC, Provo, UT 84602, hales.swift@gmail.com), Alan T. Wall (Battlespace Acoust. Branch, Wright-Patterson Air Force Base, Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Noise from high-performance military aircraft can pose challenges to community relations near airfields. Accurately predicting and quantifying community impacts is important for efforts to minimize such impacts and reduce annoyance. In this study, sound recordings measured 305 m from a tethered F-35 aircraft operating at various engine conditions are analyzed using sound quality metrics. The calculated metrics are inputs for a model of perceived annoyance used to estimate the relative contributions of loudness and other sound quality features to annoyance. These results can help inform future efforts at noise reduction by identifying potentially relevant sound quality components of the jet noise as well as helping inform discussions of noise policy on military bases. [Work supported by a USAFRL SBIR and the F-35 JPO. Distribution A: Approved for public release; distribution unlimited. Cleared 07/10/2017; JSF17-714.]

4:30

1pNS12. Spatial interpolation of noise monitor levels. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil)

Continuously recording noise monitoring stations provide feedback of the noise environment at monitor locations. While this feedback is useful, it only provides information at a few point locations, and in many cases it is of interest to know the noise level(s) at the locations between and beyond noise monitoring locations. In this study, we test the accuracy of several spatial interpolation models with experimental data collected during the Strategic Environmental Research and Development Program (SERDP) Community Attitudes Towards Military Blast Noise study. These datasets include 9 months of blast noise events captured at two different study locations. In both cases, a small number of monitors (e.g., 3–9) were located over a large region of interest (e.g., 1–8 km^2), thus providing realistic operational conditions. The utility of deterministic (e.g., nearest neighbor, Delaunay triangulation, thin plate splines, etc.) and stochastic (e.g., geostistical or kriging) interpolation models for estimating single-event and cumulative noise levels is examined using leave-one-out cross validation. The accuracy of each approach is assessed with the root-mean-square-error (RMSE), and we discuss the practical implications of implementing such approaches in real-time systems.

4:45

1pNS13. Acoustic excitation impact on aerodynamic drag measured in aeroacoustic liners. Christopher Jasinski (Univ. of Notre Dame, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com) and Thomas Corke (Univ. of Notre Dame, Notre Dame, IN)

Research interest has steadily grown for understanding the aerodynamic drag produced by acoustic liners for commercial turbofan engines. This is driven by an aim to understand the phenomena fundamentally as well as for application in flight. Stringent government regulations on aircraft noise and next generation aircraft designs that may include liners on more surfaces are key drivers for industry involvement. While the conventional perforate-over-honeycomb liner has proven effective acoustically for decades, liner drag production has not been fully understood. When an acoustic liner sample is excited with sound pressure levels above 140dB re: 20 micropascals, a

measurable drag increase is observed at flight velocity. Recent measurements have shown that tonal noise at the same level can produce more than a 50 percent increase in drag coefficient for a liner sample at lower test speeds. By testing liner samples at low speed in the Notre Dame Hessert Laboratory, detailed hotwire probe measurements near the wall have been made and drag coefficient comparisons have been made with the use of a linear air-bearing force balance. The development of the measurement setup, the results produced, and a discussion of implications will be included in this paper.

MONDAY AFTERNOON, 4 DECEMBER 2017 BALCONY I/J/K, 1:25 P.M. TO 4:00 P.M.

Session 1pPA

Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics: 30th Anniversary of the National Center for Physical Acoustics

Richard Raspet, Cochair
Univ. of Mississippi, University, MS 38677

Craig Hickey, Cochair
National Center for Physical Acoustics, University of Mississippi, 145 Hill Drive, P.O. Box 1848, University, MS 38677-1848

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Chair's Introduction—1:25

Invited Papers

1:30

1pPA1. Reflections on the origins of the National Center for Physical Acoustics. Lawrence A. Crum (APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lacuw@uw.edu)

The National Center for Physical Acoustics arose from a group of faculty within the Physics Department at the University of Mississippi that had a strong interest in the general area of physical acoustics. With Hank Bass's leadership, and support of the University of Mississippi's central administration, this group constantly sought to expand their interest and their expertise in this area. Financial support from ONR permitted the expansion of their research space and the establishment of the Institute for Technology Development in the State of Mississippi led to eventual support from Congress. In October of 1986, by an act of the Congress of the United States of America, the National Center for Physical Acoustics was established. This presentation will provide some of the history concerning this event.

1:50

1pPA2. Before there was a National Center for Physical Acoustics: Memories of the first employee. Kenneth E. Gilbert (National Ctr. for Physical Acoust., Univ. of MS, P.O. Box 35, 1703 Hunter Rd., Thaxton, MS 38871, kgilbert@olemiss.edu)

New laboratories, like newborn babies, usually come into the world along with a mix of excitement, joy, anxiety, and pain. Most of these feelings were present in the Ole Miss physics department when I arrived on the campus in August of 1985. Over the next few years, I watched what started as an attempt to obtain a metal building for acoustics research, evolve into what is today the National Center for Physical Acoustics (NCPA). In this presentation, I will tell some stories about the birth and early days of the NCPA as seen by a newly arrived outsider. Some of the stories are funny and others are not. I hope to convey what it was like to know something important was happening even though no one, especially me, seemed to know exactly where things were headed.

2:10

1pPA3. From underwater thunder to cavitation from pulsed ultrasound: One graduate student's perspective of the years leading up to the formation of NCPA. Anthony A. Atchley (Graduate Program in Acoust., Penn State Univ., College of Eng., 101C Hammond Bldg., University Park, PA 16802, atchley@psu.edu)

The Physical Acoustics Research Group at the University of Mississippi (PARGUM) fostered a vibrant, multidisciplinary research environment in the early 1980s that incubated the concept of the National Center for Physical Acoustics (NCPA). It also provided graduate students, new to physical acoustics, an enticing glimpse into the breadth of the field and an exposure to a range of important

contemporary problems. The research programs included terrestrial lightning detection and location, the underwater sound generated by lightning striking the ocean surface, long range sound propagation, propagation through porous materials, sound amplification by selective molecular excitation, acoustic levitation of bubbles and droplets, studies of acoustic cavitation thresholds and sonoluminescence, numerical simulation of bubble dynamics, and others. Beyond the scientific activities, PARGUM introduced a number of new acousticians to the ASA, many of whom have contributed to the leadership of the Society. The purpose of this presentation is to review some of the research results produced by the members of group from the time period as well as their contributions to the ASA.

2:30

1pPA4. Postdoc at NCPA: Explosive growth and entertainment. W. P. Arnott (Dept. of Phys., Univ. of Nevada Reno, Reno, NV 89557, arnottw@unr.edu)

My postdoctorate from August 1988 to December 1991 at the University of Mississippi and NCPA favorably influenced my career. Thinking back on the cast of characters and the dramatic science being pursued there brings a smile. My main mentors were Dr. James Sabatier, Dr. Richard Raspet, and Dr. Henry Bass. I was involved with theory and experiments for acoustic to seismic coupling; land mine detection; nondestructive measurements of soil properties; and fundamental analysis of Celcor extruded ceramic catalysts as materials for thermoacoustic stacks. We often worked in real-world settings—cotton and soybean fields, and at "Audi Acres" where we used massive low frequency speaker elevated with construction scaffolding and a winch as sound sources. I worked hours with Mike the machinist constructing acoustical resonators and thermoacoustic heat engines. The vibrant social and political scene was something to behold—including early morning jogging in the Faulkner woods, regular basketball games at lunch; visiting politicians and program managers, dancing on tables at the Gin; Friday nights at Dr. Bass' house; visitors from acoustic institutions around the world; Starnes catfish restaurant.

2:50–3:05 Break

3:05

1pPA5. National Center for Physical Acoustics Aeroacoustics Group. Lawrence Ukeiley (Mech. and Aerosp. Eng., Univ. of Florida, MAE-A Rm. 312, PO Box 116250, Gainesville, FL 32611, ukeiley@ufl.edu), Nathan E. Murray (NCPA, Univ. of MS, University, MS), and Bernard Jansen (NCPA, Univ. of MS, Oxford, MS)

The Aeroacoustics Group at the National Center for Physical Acoustics (NCPA) was formed in 1999 to open a new avenue of research at the center. Since its inception, the group has strived to study fundamental problems of flow-generated pressure fluctuations (both acoustic and hydrodynamic) and transition them to applied problems predominantly in the aerospace field. Over the years these problems have included vibrations in bodies traveling at supersonic speeds, jet noise associated with supersonic and subsonic exit conditions, and flow over open cavities to name a few. As part of these efforts, the group has added substantial unique infrastructure to the NCPA expanded the capabilities of the research done in house. The presentation will highlight key features of the problems, which have been studied by the group along with the key personnel.

3:25

1pPA6. Current research thrusts at NCPA. Josh R. Gladden (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

Over the 30 year history of the National Center for Physical Acoustics, the technologies have advanced and research has evolved; however, the mission has remained the same: to pursue solutions to practical problems using a fundamental physics approach, to educate the next generation of physical acousticians, and to serve as a translational bridge between basic research generated by our academic communities and the private sector. In this talk, I will discuss major research thrusts and accomplishments of NCPA in the last 5–8 years. Topical areas covered will be infrasound and long range propagation, aeroacoustics, structural health monitoring, physical ultrasound, and acoustics in porous media. I'll include some discussion of prominent scientists driving this research along with a few funny stories.

Contributed Paper

3:45

1pPA7. The importance of tuning curves and two-tone tests in nonlinear acoustic landmine detection. Miahnna Nguyen and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m204578@usna.edu)

During the 1990s through ~2010, Jim Sabatier's group (Univ. Mississippi and NCPA) had made significant progress in understanding airborne acoustic landmine detection. In particular, using their development of scanning laser Doppler vibrometry, plastic anti-tank landmines could be detected in soil along roadbeds. Research on the nonlinear acoustic detection of buried landmines by Dimitry Donskoy (Stevens Tech) lead to lumped element models of the target which included a nonlinear spring-like mechanism. Tom Muir studied seismic excitations. Sabatier and Korman suggested that the soil nonlinearity and the soil interaction with the flexural vibration of the top-plate of the landmine or drum-like simulant were both significant factors in nonlinear landmine detection. Nonlinear tuning curves of the soil vibration response vs. frequency (directly over the target) along with rich combination frequency generation in two-tone tests suggested that the vibration was similar to Paul Johnson's group results on mesoscopic nonlinear elasticity found in geomaterials. Recent airborne acoustic nonlinear landmine detection experiments compare drum-like simulant tuning curves to results using a soil plate oscillator apparatus (two circular flanges sandwiching and clamping a thin circular elastic plate that supports a cylindrical level column of sand above the plate) to help model flexural mesoscopic nonlinear hysteretic behavior.

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration

Benjamin Shafer, Chair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Contributed Papers

3:30

1pSA1. Generalized control metric for active structural acoustic control: A novel understanding for controlling symmetric structures. Yin Cao and Hongling Sun (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., Qing Nian Hui Jia Yuan Bldg. 6, No. 411, Beijing 100022, China, caoyfive@gmail.com)

Active structural acoustic control (ASAC) is a strategy aiming at controlling the radiated sound of the structure by controlling the vibration. Generally, controlling the vibration is not the same as controlling the radiated sound. Therefore, the choice of the control metric also referred to as the cost function in ASAC is a very important research topic and has been received a lot of attentions by researchers. Previous research has identified a novel control metric named the weighted sum of spatial gradients (WSSG). An analysis has shown that WSSG is able to closely approximates the performance of using radiated sound power as the control objective by choosing the right weights. In this paper, based on previous findings, a generalized control metric of active structural acoustic control for symmetric structures is discussed and analyzed. For symmetric structures, the radiated resistance matrix is always bisymmetric. It will be shown that controlling the vibration is able to achieve the performance of directly controlling the radiated sound power provided that the secondary plant matrix is overdetermined. A uniformed control metric for active structural acoustic control of symmetric structures will be presented and analyzed.

3:45

1pSA2. Experimental study of the impact noise through concrete floor with resilient layers. Abdelouahab Bouttout (DPBE, CNERIB, National Ctr. of Studies and Integrated Res. of Bldg. Eng., CNERIB,Cité Nouvelle El-Mokrani, Souidania, Algeria, Souidania 16097, Algeria, Bouttout@gmail.com)

In this paper, we present an experimental approach to evaluate the sound transmission through floors and building structures. This method is developed in our laboratory at the National Center of Studies and Integrated Research of Building Engineering, CNERIB, Algeria. The method is based on measurement of sound due to impact vibration generated by a standard tapping machine placed on the upper surface of a small concrete floor. This floor has dimensions of 1.2(m)×0.8(m)×0.2(m) and simply supported on four elastic supports with 2 cm of rubber layer. The natural frequency of the floor is measured with an ambient vibration apparatus, and the obtained value is $f = 19$ Hz. The measurement consists of recording vibration levels in the upper and lower surfaces of the floor, in 1/3 octave frequency bands between 16 Hz and 16 KHz using piezoelectric accelerometer Brüel and Kjær 4508. The sound level meter Brüel and Kjær 2270 has been used to analyze the vibration signal for different cases. The weighted sound reduction index of the floor has 4 dB without any cover. Some samples of cover floor are tested using this procedure. The results show that the weighted sound reduction index of resilient layer can reach 18 dB, 11.6 dB and 6.6 dB for the rubber, Bitumen and PVC material, respectively. The important results obtained in this paper can be used as a platform to correct the impact sound insulation in multi-storey residential building and renovation plans using recycled resilient materials.

4:00

1pSA3. Prediction of ground-borne vibrations due to elevated railway traffic. Salih Alan and Mehmet Caliskan (Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

This study investigates the impact of ground-borne vibrations on people in dwellings nearby elevated railway traffic. Assessments are conducted over predicted vibrations with respect to national regulations. Main excitation mechanism is taken as the dynamic loading due to rail and wheel irregularities. Frequency response functions at the rail head are obtained from harmonic analyses by finite element modeling of the elevated structure. In the prediction procedure, dynamic model of the vehicle is coupled with these frequency response functions. Dynamic loading on the railhead is determined and dynamic reaction forces on the legs of the elevated structure are calculated in frequency domain. Ground vibrations are then estimated by implementing a Fourier transform based theoretical model for the layered ground. Assessment in the frequency range from 1 Hz to 100 Hz in one-third octave bands are conducted for groundborne vibrations calculated at the foundation level of dwellings.

4:15

1pSA4. Vibrometric characterization of a TN-32 dry storage cask for spent nuclear fuel. Kevin Y. Lin (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., PO BOX 1848, Oxford, MS 38677-1848, klin@go.olemiss.edu), Wayne E. Prather, Zhiqu Lu, Joel Mobley, and Josh R. Gladden (National Ctr. for Physical Acoust., Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

The assessment of the internal structural integrity of dry storage casks for used high burnup nuclear fuel assemblies is of critical importance before these are transported to permanent repositories. The size of the casks (5.2 m in height and 2.4 m in diameter), structural complexity, and the inability to access the interior make this a challenging task. This project addresses these difficulties through a multi-modal approach involving nuclear, charged particle, and acoustic methods. In this talk, we report on linear and nonlinear vibrational spectra of intact TN-32 casks. These studies use both impulsive and swept continuous-wave excitations with a variety of sensor placement configurations. From the resulting spectra, resonant frequencies, quality factors, and harmonic responses of various vibrational modes were determined. A detailed finite element model of the TN-32 was constructed and the experimental results are compared to the modal structure determined numerically. [This work was supported by DOE NEUP Award: DENE0008400.]

4:30

1pSA5. The determination of mode shapes of a scaled model TN-32 spent nuclear fuel dry storage cask. Kevin Y. Lin (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., PO Box 1848, Oxford, MS 38677-1848, klin@go.olemiss.edu), Wayne E. Prather, Zhiqu Lu, Joel Mobley, and Josh R. Gladden (National Ctr. for Physical Acoust., Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

The assessment of the internal structural integrity of dry storage casks for used high burnup nuclear fuel assemblies is of critical importance before

these are transported to permanent repositories. The size of the casks (5.2 m in height and 2.4 m in diameter), structural complexity and the inability to access the interior make this a challenging task. This project addresses these difficulties through a multi-modal approach involving nuclear, charged particle and acoustic methods. In this work, we report on measurements of the vibrational spectra and mode shapes using a 6:1 scaled TN-32 model cask constructed in our lab. 2-D vibrometric scans were performed with a laser Doppler velocimeter to measure the vibrational mode shapes exhibited by the cask. Good agreement is observed between the experimentally measured mode shapes and those determined numerically using a detailed finite element model. Several modes are identified that will be important in assessing the internal characteristics from external measurements. [This work was supported by DOE NEUP Award: DENE0008400.]

MONDAY AFTERNOON, 4 DECEMBER 2017 ACADIA, 2:00 P.M. TO 5:00 P.M.

Session 1pSC

Speech Communication: Voice, Tone, and Intonation (Poster Session)

Cynthia P. Blanco, Chair
Department of Psychology, Northwestern University, 2029 Sheridan Road, Evanston, IL 60208

All posters will be on display from 2:00 p.m. to 5:00 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 2:00 p.m. to 3:30 p.m. and authors of even-numbered papers will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

1pSC1. Differential pitch analysis for tone and accent languages. Ettien N. Koffi (English, Saint Cloud State Univ., 720 Fourth Ave. South, Saint Cloud, MN 56301, enkoffi@stcloudstate.edu)

According to the Critical Band Theory, the auditory perception of F0 data is the same for all human beings. However, when F0 signals are transferred through the auditory cortex to specialized areas of the brain, they are perceived and processed differently depending on whether the language is tonal or accentual. In tone languages, F0 data appears to be processed in Heschl's gyrus (Schneider 2005), whereas in accent languages, it is processed via the thalamus to inferior frontal regions (Myers 2017). Furthermore, in accent languages, F0 signals are computed on a nominal scale, but in tone languages, an ordinal scale is used (Speaks 2005). These insights support the long-held linguistic view that accent and tone languages are prosodically different. Terms such as strong/weak, stressed/unstressed are used to describe pitch variations in accent languages, whereas in tone languages, the terms used are extra low, low, mid, high, and extra high. Due to these prosodic and processing differences, it is not advisable to apply the same interpretive framework in analyzing pitch variations in tone and accent languages. Examples will be provided from English and Anyi, a West African language, to underscore the pitfalls in doing so.

2:00

1pSC2. What happens when f0 movements and prosodic units were randomly aligned? Wei Lai, Nari Rhee, and Mark Liberman (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, weilai@sas.upenn.edu)

According to intonation phonology, surface f0 is generated from superimposition of local f0 events associated with hierarchical prosodic units (e.g., Pierrehumbert, 1980). This study conducted two experiments to investigate whether or to what extent the misalignment between f0 patterns and prosodic units would result in awkward-sounding English intonation. In Experiment 1, f0 contours of utterances from Obama Weekly Address were extracted via Praat, interpolated and smoothed using a quadratic spine, and superimposed to another Obama-Weekly-Address stretch by PSOLA algorithm. The perception result, surprisingly, showed that the resynthesized sentence still sounded quite natural in intonation, regardless of the random alignment between f0 patterns and the location of stresses and phrases. In Experiment 2, the Obama-Weekly-Address stretches were superimposed with f0 contours of three different languages that differ in the size of basic prosodic domains: Mandarin (a tone language), Korean (an accent phrase language) and Ukrainian (a stress language). The perception results showed that utterances superimposed with Korean and Ukrainian f0 contours still sounded quite natural and native, while Mandarin f0 contours sounded somehow fluctuational on English segments. These results suggested that f0 movements might not be largely determined by prosody structures, but by language-independent constrains such as physiological plausibility and expediency.

1pSC3. Native perception of Cantonese tones in high-variability conditions. Yan Chen (Linguist Dept., Univ. of Arizona, TUCSON, AZ 85721, yanchen@email.arizona.edu)

This study investigates how native speakers of Cantonese perceive and produce Cantonese tone pairs T2-T5, T3-T6, T4-T5, T4-T6, and T5-T6 in high-variability conditions. Thirty five native speakers of Cantonese from Hong Kong participated in a 6AFC identification task (screening test), a mixed-talker AXB task (ISI = 1500 ms), and a mixed-talker repetition task (delay = 1500 ms). Results from the screening test showed that the participants did not merge the tones examined in this study. Results from the AXB task showed that (1) perception of tones with f0-height differences (T2-T5 and T3-T6) was more challenging than perception of tones with f0-contour differences (T4-T5, T4-T6, and T5-T6), (2) tones contrasting in direction of f0 change (T4-T5) were the easiest, and (3) perception of level tones (T3-T6) was more difficult than perception of contour tones with an f0-height difference (T2-T5). Data from the repetition task were fitted with quadratic polynomial equations ($y = a + bx + cx^2$) and the analysis revealed that the participants produced the two level tones—T3 and T6—distinctively (in terms of coefficient-a) and the two rising tones—T2 and T5—distinctively (in terms of coefficient-c). This indicates that the participants perceived different level tones and different rising tones in the repetition task (i.e., no tone merger).

1pSC4. English focus prosody processing and production by Mandarin speakers. Chikako Takahashi, Hyunah Baek, Sophia Kao, Alex. H. L. Yeung, Marie K. Huffman, Ellen Broselow (Dept. of Linguist, Stony Brook Univ., SUNY at Stony Brook, Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu), and Jiwon Hwang (Asian and Asian American Studies, Linguist, Stony Brook, NY)

Our study compared the processing and production of English focus prosody by native speakers of English and Mandarin. Twenty-one Mandarin speakers living in the US and 21 English speakers participated in two tasks. In the processing task, participants responded to instructions that contained natural or unnatural contrastive prosody (Click on the purple sweater; Now click on the SCARLET sweater/Now click on the PURPLE jacket.) In the production task, participants guided an experimenter to place colored objects on a white board, with some contexts designed to elicit contrastive focus (Put the yellow arrow over the ORANGE arrow/yellow DIAMOND, please). All adjectives and nouns were bisyllabic trochees. The two groups differed in their realization of focus, with English speakers tending to align the pitch peak with the stressed syllable and Mandarin speakers with the right edge of the focused word. However, comparison of reaction times for the processing task indicated that both groups responded more quickly to instructions with natural than unnatural prosody, although English speakers' response times were significantly faster in both conditions. We argue that although Mandarin speakers show Mandarin-like realization of focus in their production, they can nonetheless use the English prosodic patterns in their processing.

1pSC5. Towards a model of Tatar intonational phonology. Adam J. Royer (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, ajroyer@ucla.edu)

This study is a preliminary report of our ongoing research investigating an Autosegmental-Metrical model of intonational phonology of Kazan Tatar, a Turkic language spoken in Tatarstan, Russia. Tonal patterns of neutral focus utterances were examined by varying the length of words and phrases, the location of stresses, syntactic structures, and sentence types. Results suggest that Tatar has three prosodic units marked by intonation. They are the Accentual Phrase (AP), the Intermediate Phrase (ip), and the Intonational Phrase (IP). An AP has one or more words and has an optional initial high tone (Hi), realized at the left edge or leftmost stressed syllable of the AP, and an obligatory rising pitch accent (L+H*) aligned with the rightmost stressed syllable of the AP. Interestingly, the unstressed syllables after the pitch accented syllable show high f0 until the end of the AP, suggesting an optional AP-final H tone (Ha). An ip is marked by a phrase-final high tone (H-) realized on an ip-final syllable, which is slightly longer than AP-final syllable. Finally, an IP is marked by a phrase-final boundary tone realized on a substantially lengthened syllable. A fuller model of Tatar will be compared with the intonation models of other Turkic languages.

1pSC6. The nature of variation in the tone Sandhi patterns of Wuxi Wu. Hanbo Yan (School of Chinese Studies and Exchange, Shanghai International Studies Univ., 550 West Dalian Rd., Bldg. 2, Rm. 418, Shanghai, Shanghai 200083, China, yanhanbo@shisu.edu.cn) and Jie Zhang (Linguist, Univ. of Kansas, Lawrence, KS)

The Northern Wu Chinese dialect of Wuxi has two different tone patterns in disyllables—a pattern with tone sandhi that involves a synchronic chain-shift and a no sandhi pattern, and the two patterns apply variably. This study investigates the nature of this variation. Seventy-one native speakers participated in three rating experiments that investigated disyllables "subjective frequency, semantic transparency, and the variant forms" perceptual goodness. Results show that modifier + noun combinations prefer the sandhi form more than verb + noun combinations. Lexical frequency has a positive effect on sandhi application in both modifier + noun and verb + noun items. Semantically transparent items are less likely to undergo tone sandhi, but only for verb + noun combinations. These results are interpreted with respect to the properties of wordhood in Wuxi and Chinese dialects in general and the productivity of the chain-shift tone sandhi pattern observed in an earlier production study.

1pSC7. Eye movements and speech prosody in the processing of information structure: An exploratory study in Mandarin Chinese. Li Liu, Ying Chen, and Xueqin Zhao (English, School of Foreign Studies, Nanjing Univ. of Sci. and Technol., 200 Xiaolingwei St., Nanjing, Jiangsu 210094, China, ychen@njust.edu.cn)

An exploratory study on information processing in Mandarin discourse was carried out using the remote system of eye tracker. A map was designed to display four destination images, four written words of orientations and four values of distance from the map center to the destinations. Participants answered five pre-recorded questions while reading the map on the computer screen in three trials. The experimenter pretended not to have recorded Trial 1 and requested the participants to do the task again in Trial 2. The questions in Trial 3 were recorded with a different gender to indicate a new inquirer. The preliminary results show some major tendencies: (1) gaze points and saccade routes in Trials 2 and 3 were fewer and less complicated compared to Trial 1; (2) fixation duration on the target words was shorter when the information was old than when it was new; (3) duration, F0 and intensity of the target words were reduced in Trials 2 and 3 compared to Trial 1; (4) the duration tended to be longer, the F0 higher but the intensity weaker in Trial 3 than in Trial 2, suggesting a gender difference in the speech, which was also reflected in the eye-movement data.

1pSC8. Prosodic asymmetry in phonetic reorganization of Seoul Korean 3-way voiceless stop contrast. Yoonjeong Lee and Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, yoonjeol@usc.edu)

In younger generation Seoul Korean speakers, a phonetic reorganization of VOT and f0 in the phrase-initial stop contrast (i.e. aspirated, lenis, fortis) has been well documented. The current study elucidates how this local consonant effect on f0 further interacts with the global tonal patterns of the accentual phrase (AP). We test how the initial 3-way stop contrast is phonetically realized in AP-initial and AP-internal prosodic positions for six younger generation speakers (born 1980-1990). Our results show that the consonant effect on f0 is *categorical* in AP-initial position, compared to exhibiting a *gradient* effect in AP-internal position. We confirm an AP-initial VOT merger between aspirated and lenis stops, accompanied by an increased between-consonant f0 difference. In AP-internal position, along with a small but significant f0 difference, we found a near-merger of VOT between lenis and fortis stops, arising from the substantially reduced occurrence of intervocalic lenis voicing. Taken together, our findings provide novel evidence of an intricate interaction between phrase-level prosody and the local phonetic reorganization in the newly emerged phonetic system of Seoul Korean stops. [Work supported by NIH.]

1pSC9. Acoustic similarities among voices. Part 2: Male speakers. Jody E. Kreiman, Patricia Keating, and Neda Vesselinova (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Little is known about how to characterize normal variability in voice quality within and across utterances from normal speakers. Our previous study of female voices suggested that only a few acoustic parameters consistently distinguish speakers, with most of the work being done by idiosyncratic subsets of parameters. The present study extends this research to samples of 50 men's voices. The men read 5 sentences twice on 3 days—30 sentences per speaker. The VoiceSauce analysis program estimated means and standard deviations for many acoustic parameters (including F0, harmonic amplitude differences, harmonic-to-noise ratios, and formant frequencies) for the vowels and approximant consonants in each sentence. Multidimensional scaling and linear discriminant analysis were used to examine the acoustic characteristics of the overall voice space, and to measure how well each speaker's set of 30 sentences could be acoustically distinguished from all other speakers' sentences. Additional analyses of small subsets of voices compared the importance of global versus local details in discriminating voices acoustically. Results will be compared to those for female speakers, and implications for recognition by listening will be discussed. [Work supported by NSF and NIH.]

1pSC10. Voice quality variation over the course of the English utterance. Elizabeth M. Bird (BioEng., UCSD, 9500 Gilman Dr, 9252 Regents Rd., E, La Jolla, CA 92093, embird@ucsd.edu) and Marc Garellek (Linguist, UCSD, La Jolla, CA)

The presence of atypical voice quality (e.g. breathy and creaky voice) can be used to diagnose voice disorders, but it is also clear that healthy English speakers' voices vary as a function of phrasing. For example, ends of declarative utterances tend to be produced with creakier voice quality. Yet for speakers who have no diagnosed voice disorder, it is still unclear what factors affect voice variation over an utterance, and how consistent these variations are across speakers. This in turn limits diagnostic utility of perceived non-modal voice. Therefore, the goal of this project is to determine how the voice varies over the course of English utterances. We recorded electroglottographic (EGG) and audio waveforms of 20 male and female speakers of Californian English reading sentences of various lengths that were designed to avoid non-modal voice associated with certain segments. The EGG waveforms were analyzed for contact quotient and f0 at the beginning, middle, and end of the utterance. Discussion will focus on how voice quality differs by position in utterance, and the extent to which it is predictable from f0. We also discuss whether voice quality varies as a function of gender, utterance length, and speakers' average voice quality.

1pSC11. Live versus art in voice change: The case of Maria Callas. Nina Eidsheim and Jody E. Kreiman (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Many commentaries on the voice of singer Maria Callas note that her voice changed markedly over the course of her career, with changes often attributed to "ferocious dieting." Such claims are particularly troubling in the absence of evidence that weight loss affects voice acoustics, and in the relative absence of acoustic data testing specific hypotheses regarding expected changes in voice with dieting. This paper examines recordings from early and late in Callas's career, and attempts to determine whether observed changes are more consistent with the acoustic effects of physiological changes associated with extreme and rapid weight loss (changes in hormone levels, respiratory changes, differences in tongue size/vocal tract dimensions, reflux, etc.), with aging (e.g., increasing vocal instability, changes in resonance frequencies, changes in F0), or with artistic choices.

1pSC12. Phonatory characteristics of *ex-vivo* aged sheep models. Michael Döllinger (Div. for Phoniatrics and Pediatric Audiol. at the ENT Dept., Univ. Hospital Erlangen, Bohlenplatz, 21, Erlangen, Bavaraia 91054, Germany, michael.doellinger@uk-erlangen.de), Markus Gugatschka (ENT University Hospital, Dept. of Phoniatrics, Medical Univ. of Graz, Graz, Austria), Olaf Wendler (Dept. of Otorhinolaryngology, Head and Neck Surgery, Experimental ENT Res. Lab. I, Univ. Hospital Erlangen, Erlangen, Bavaria, Germany), Claus Gerstenberger (ENT University Hospital, Dept. of Phoniatrics, Medical Univ. of Graz, Graz, Austria), Hossein Sadeghi, and Stefan Kniesburges (Div. for Phoniatrics and Pediatric Audiol. at the ENT Dept., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

For humans, voice quality reduces with age. Due to the increasing life span and the increase of the older population, it is necessary to develop new strategies to treat people concerned. Since potential new treatment technologies as functional electrical stimulation have to be first established in animal models, we investigated phonatory characteristics in aged sheep larynges. Ex-vivo dynamic experiments were performed on nine aged sheep larynges providing normative phonatory data. The larynges were analyzed at sustained phonation for varying subglottal pressure levels. Additionally, three different weights were successively attached to the thyroid cartilage to induce prestress forces towards the thyroarytenoid muscle and to therefore simulate longitudinal tension of the vocal folds. Laryngeal vibrations, airflow and acoustics were recorded. Afterwards, larynges were analyzed for tissue damage using histological standard methods. Overall, the larynges showed rather asymmetric vibrations and exhibited very soft and pliable vocal fold tissue characteristics. The histological analysis showed (in two cases) minimal damage in form of slight epithelial layer detachments and smaller cracks in the lamina propria. Further and detailed results will be presented. Next, the gathered data and characteristics will be compared to young sheep as well as to aged sheep being treated with functional electrical stimulation.

1pSC13. Comparing roughness in sustained phonations and connected speech using a matching task. Supraja Anand, David A. Eddins (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, suprajaanand@usf.edu), and Rahul Shrivastav (Office of the Vice President for Instruction, Univ. of Georgia, Athens, GA)

Experiments on perception of dysphonic voice quality have typically relied on sustained vowel phonations. The strong preference for vowels could be explained by their ease of production, their time invariance, and by the absence of confounding articulatory (e.g., consonant, dialect) and prosodic (e.g., stress, rate) changes. However, the magnitude of roughness perceived from sustained vowels may not reflect the roughness in connected speech produced by the same speaker. We examined how stimulus type impacts the perception of roughness. Ten naïve listeners judged roughness for both vowel /a/ and sentences selected from ten dysphonic speakers using a single-variable matching task. Stimuli were selected to ensure a continuum of vocal roughness for vowels. The intra- and inter-listener reliability estimated using intra-class correlation (ICC) as well as matching values and variability will be compared across each stimulus type. This work is essential for extending models of dysphonic voice quality perception from vowels to connected speech, which are likely to correspond more closely with perceived handicap and better represent relevant treatment targets for patients with dysphonia.

1pSC14. Measurement scales, psychophysical models, and reliability statistics in voice quality perception. Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, French East 344, 3202 Eden Ave., Cincinnati, OH 45267, silbernh@ucmail.uc.edu)

Perceptual judgments of voice quality can be given on discrete or continuous scales. Discrete scales are typically described as consisting of "equal appearing intervals," and continuous judgments may be given on visual analog scales (VAS) or via direct magnitude estimation. Interrater agreement on discrete scales is highly variable and often poor, though there is evidence that application of psychophysical principles (i.e., perceptual variability, response bias) can increase reliability. There is some evidence that interrater reliability may be higher with continuous scales, though the influences of psychophysical assumptions and measurement properties on the reliability of continuous scales has not been analyzed particularly thoroughly. The present work considers the effects of different assumptions about underlying perceptual and decisional processes on the measurement properties and reliability of perceptual voice quality scales. An adaptation of Stevens' power law combined with different decision rules provide a model of the mapping between the acoustic properties known to influence voice quality (e.g., H1-H2, HNR) and discrete or continuous clinical judgment scales (e.g., "breathiness," "roughness"). The effects of variation in model parameters on reliability and agreement statistics will be analyzed, and novel agreement statistics based on the properties of ratio-scale judgments will be explored.

1pSC15. On the restoration of Shackleton's voice. SangHwi Jee and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, Seoul 06978, South Korea, slayernights@ssu.ac.kr)

Shackleton's voice, which attempted to explore Antarctica 100 years ago, was vividly restored with voice-processing technology. The restored voice was used as a restoration criterion after restoring the sound quality to a sound quality by applying noise processing technology to the voice of Shackleton who finished the exploration and gave a speech at a university. The basic voices of the voices are similar to those of family members and relatives because of the similarity of the gene structure and the body structure. Using Shackleton's great-grandson voices, he restored Shackleton's voice by adjusting the rate of speech, intensity, intonation, duration, tone changes, and tone averages as analyzed in Shackleton's voice. The subjective evaluation of the restored voice showed a similarity of more than 90% in the spectral SNR compared to the original voice, and when MOS evaluation was performed by the linguistic expert, 4.5 / 5 was obtained and the restoration of the voice of Shackleton was successful respectively. It is expected that the restoration technology of the deceased using the relative of relatives can restore the message of love and encouragement to the family in their future voice to the descendants.

1pSC16. Individual differences in the signaling of prosodic structure by changes in voice quality. Stefanie Shattuck-Hufnagel (MIT, 36-511 MIT, 77 Mass Ave., Cambridge, MA 02139, sshuf@mit.edu)

Earlier work has shown that speakers of American English often (although not always) produce irregular pitch periods (or other changes in voice quality) at prosodically significant locations, such as the onset of a new intonational phrase or a pitch-accented syllable, when those constituents begin with a [+voiced] phonemic segment (Pierrehumbert & Talkin 1992). This tendency varied substantially across 5 speakers of FM-Radio-News-style speech (Dilley *et al.* 1996). Further work with speakers of a west-coast dialect (Garellek 2012) raised questions about the appearance of this cue at phrase onsets, when the phrase begins with a prosodically weak (lexically unstressed) syllable; this suggests that it occurs only for phrasally-strong (pitch accented) syllables, whether they occur in phrase-initial position or not. In contrast, extensive analysis of voice quality changes in a corpus of imitated utterances produced by speakers of a mid-western dialect (Cole & Shattuck-Hufnagel 2011) clearly shows a tendency for voice-quality changes at the onsets of prosodically weak syllables when they occur at the onsets of intonational phrases. This raises the possibility that individual speakers or dialects may differ in the likelihood of using changes in voice-quality to mark different types of prosodic events, such as phrase onsets vs pitch accents.

1pSC17. Delayed auditory feedback induces slow oscillations in vocal fundamental frequency and intensity. Jieun Lee and Francois-Xavier Brajot (Commun. Sci. and Disord., Ohio Univ., Gover Ctr. W218, Athens, OH 45701, jl976314@ohio.edu)

A system with negative feedback tends toward oscillation or instability when delay is introduced into the feedback loop. Given current evidence for feedback control of pitch and loudness, we hypothesized that delaying auditory feedback during sustained phonation would result in oscillations in fundamental frequency and vocal intensity. Subject produced a sustained vowel across seven auditory feedback conditions: 0, 100, 200, 300, 400, 500, 600, and 700 ms delay. Increasingly large oscillations in fundamental frequency and intensity were observed as delay increased. The frequency of oscillations as a function of delay appear to be subject-specific, with a range between 0.3 and 0.8 Hz. These oscillations carried over into the prosodic contour of subjects' connected speech produced under the same delay conditions. These results suggest that disfluencies induced by delayed feedback (e.g., sound prolongations) may be the result of compensation for shifts in pitch and loudness, rather than discrepancies in articulatory or segmental timing.

1pSC18. Volume velocity field and acoustics measurements in a canine larynx model. Charles P. Farbos de Luzan, Liran Oren, Sid M. Khosla, Alexandra Maddox (Dept. of Otolaryngol. - HNS, Univ. of Cincinnati, Medical Sci. Bldg., Rm. 6303B, Cincinnati, OH 45267-0528, farboscs@ucmail.uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

In the classic source-filter theory, sound is produced at the glottis by a process known as flow modulation; in this case, "flow" specifically refers to the flow rate (Q) produced at the glottal exit during the phonation cycle. Flow modulation refers to the fact that Q is changing as the glottis opens and closes dQ/dt. Although dQ/dt is constantly changing from glottal opening and closing, the greatest rate of change happens during the latter part of closing, when Q rapidly decreases. This rapid deceleration is quantified by the maximum flow declination rate (MFDR). MFDR has been shown to highly correlate with acoustic intensity (loudness). The aim of this study is to measure changes in Q and the acoustic energy in a vibrating canine larynx model as a function of subglottal pressure and vocal tract constriction. Volume flow measurements are taken using time-resolved tomographic-PIV. Q at the glottal exit is extracted from PIV measurements and MFDR is computed from the waveform. Acoustic measurements (SPL) are taken simultaneously. Testing is done with and without vocal tract, which is place above the larynx. The constriction in the vocal tract is varied by changing the distance between the 2 false vocal folds (FVF). Each case above is tested at low and high subglottal pressures. Measurements show that the glottal exit flow is complex. The waveform of Q is skewed towards closing phase, even without a vocal tract. The skewing is affected further by the vocal tract constriction. Both MFDR and SPL increase with subglottal pressure.

1pSC19. Cooperation via communication: Influencing vocal alignment in conversation. Elliot Pollack and Elizabeth D. Casserly (Psych., Trinity College, 300 Summit St., Box 702353, Hartford, CT 06106, elliot.pollack@trincoll.edu)

Alignment of human vocal behavior is a well-documented phenomenon, however, the factors which influence its direction and magnitude are not firmly established. Components of speech which are subject to alignment include—but are not limited to—word choice, syntax, and rate of speech. In the present study, Speakers completed a puzzle task which required them to communicate with a Model whose fundamental frequency (F0) changed during the interaction. Additionally, naïve subjects (Listeners) assessed the overall similarity between the Speaker's speech over time and the Model using an AXB paradigm. Speakers were found to deviate from the Model in F0, however, were perceived by Listeners to mimic the Model in a holistic measure. Speakers were rated as becoming more like the Model when this partner diverged as opposed to converged. A personality factor survey showed that greater Openness predicted less perceived similarity. The discrepancy between divergence in the acoustic measure and convergence in the perceptual measure reveals a potential hierarchy of speech factors that we use to assess alignment.

Session 1pSP

Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics: Source Tracking with Microphone/Hydrophone Arrays II

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Invited Papers

1:00

1pSP1. Direction finding and inverse imaging with microphone arrays in a forest. Michael J. White, Michelle E. Swearingen, and Patrick J. Guertin (US Army ERDC/CERL, PO Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil)

Beneath the tree canopy, wind and temperature gradients are reduced, and so are wind noise and atmospheric fluctuations that cause acoustic scintillation. However, the large number of discrete scatting objects complicates the received signal in a repeatable manner. We placed three small arrays in a forest and operated a propane cannon in the area nearby to investigate the spatial and angular dependence of the received field and its correlations. As expected, scattering from trunks impairs the ability to find the source using the conventional beamformer. Several direction finding techniques were employed to evaluate the signal direction in this complex environment. Methods are compared against both measured data and a numerical scattering model for the same location.

1:20

1pSP2. Direction of arrival estimation using local series expansion evaluated on acoustic data. Gustaf Hendeby (Elec. Eng., Linköping Univ., LiU, Linköping, N/A 581 83, Sweden, gustaf.hendeby@liu.se), Thomas Lunner (Tech. Audiol., Dept. of Experimental and Clinical Res., Linköping Univ., Snekkersten, Denmark), and Fredrik Gustafsson (Elec. Eng., Linköping Univ., Linköping, N/A, Sweden)

Direction of arrival (DOA) estimation using a local series expansion of the signal is evaluated on acoustic data. It was recently proposed that DOA estimation using a sensor array could be performed using a local temporal/spatial series expansion. That is, to use the local phase of the signal rather than its estimated second order moments or correlation as in classical methods. The approach should provide easier handling of general sensor configurations, ability to deal with very low sampling rates and few samples, and benefits from miniature sensor configurations. Additionally, the signal is not required to be narrow-banded, as sufficient smoothness is enough. Here, the local series expansion method is benchmarked to a standard beam-forming method using experimental data, recorded in an anechoic chamber at the Technical University of Denmark (DTU). The dataset was recorded using 52 high quality microphones distributed on a sphere with radius 50 mm, and contains several recordings from one or two well defined sources from several different directions, with different levels of reverberation. The dataset is used to evaluate important aspects of the new method and its potential to improve on standard DOA methods.

1:40

1pSP3. Detection of reflections in different room acoustical conditions using generalized cross correlation phase transform. HyunIn Jo, Jong Gak Seo, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

A framework for localization of the received source emission was implemented in enclosures based on generalized cross correlation (GCC) weighting with phase transform (PHAT) function. It was to achieve the goal of capturing meaningful sounds including speech in actual environments. The effects of room reverberation and ambient noise on the performance of the system were investigated with a linear microphone array. Experiments were performed in different environments with different RT and background noise levels. The sound sources are consisting of pink noise and speech sources were used to observe the relationship between localization accuracy and room acoustical parameters. Results for evaluation indicated that the location of the reflections was noticeable for accurately detection of the sources in reverberant and noisy environments.

2:00

1pSP4. Array element localization using multidimensional scaling (MDS) and matrix completion, with comparisons to more conventional non-linear least squares optimization. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

Multidimensional scaling (MDS) is an approach from the social sciences which was invented to project qualitative notions of distance onto 2D or 3D coordinate axes to provide visual displays for clustering purposes. As it turns out, this provides a viable algorithm for array element localization. When we calibrate an array using acoustic or RF ranging signals we measure inter-element distances and form a Euclidean Distance Matrix (EDM), and then convert them to 3D orthogonal coordinates. MDS converts the EDM to AEL coordinates. This process has some drawbacks. We may not get all the inter-element distances, because our ranging signals may have limited detection ranges, or because of obstructions to the line of sight. Also, it is not clear how to incorporate knowledge of unequal measurement errors among the EDM entries. Matrix completion is an approach from compressed sensing that allows us to perform the MDS paradigm, despite missing values in the EDM. We will present simulations for typical AEL scenarios and compare AEL results from MDS and matrix completion with results obtained using non-linear least squares optimization.

2:20

1pSP5. Subarray processing with coprime and minimum redundancy arrays. Guen-Soo Jo and Jung-Woo Choi (School of Elec. Eng., KAIST, Daehak-ro 291 KAIST, LG Innovation Hall(N24) 4106, Daejeon KS015, South Korea, thig1399@kaist.ac.kr)

A coprime sensor array (CSA) refers to a non-uniform linear array consisting of two undersampled uniform linear arrays (ULAs) with coprime numbers of elements, respectively. Its beam pattern can be enhanced by multiplying beampatterns of individual subarrays. This technique, called product processing, provides improved resolution because its sum coarray length is extended from the product of beampatterns. Nevertheless, the extended coarray inevitably includes holes that result in higher peak sidelobes than the full ULA case. In this paper, we propose a technique to reduce peak sidelobes through the combination with minimum redundancy arrays (MRAs) and an additional subarray. Two ULAs comprising the coprime array are thinned by the MRA technique, and the reduced number of sensors is then utilized to construct the third subarray. The third array suppresses peak sidelobes by placing nulls at the peak sidelobe locations. The characteristics of the proposed technique are analyzed in terms of white noise gain (WNG) and maximum sidelobe level (MSL). It is demonstrated that the proposed technique can improve the MSL with similar number of sensors, thereby reducing the estimation error in the presence of multiple sources.

2:40–2:55 Break

2:55

1pSP6. Recording and post-processing speech signals from magnetic resonance imaging experiments. Juha Kuortti and Jarmo Malinen (Dept. of Mathematics and Systems Anal., Aalto Univ., P.O. Box 11100, Helsinki FI-00076, Finland, juha.kuortti@aalto.fi)

Speech recordings during an Magnetic Resonance Imaging (MRI) experiment yield valuable but noisy acoustic data for modelling purposes. Despite recent improvements in optical microphones for MRI, using an acoustic sound recording system in dipole configuration, based on shielded electret microphones and waveguides, has some inherent advantages: For example, the bandwidth can be made very wide, and the extremely linear behaviour of all components facilitates the numerical post-processing of signals. Here, the full measurement chain and the critical design decisions are reviewed. In particular, one must take into account the resonant behaviour of the necessarily non-optimally terminated waveguides as well as the environment acoustics within the MRI coils. Two competing approaches for the signal post-processing were developed: (i) Optimal subtraction of the noise and speech signals, and (ii) spectral peak removal by adaptively fitted comb filters. The main objective is the high-quality spectral identification, estimation, and classification of source features, e.g., vowel formants. Approach (i) preserves the phase behaviour of the original signal. Approach (ii) produces excellent speech sound quality when the noise consists of only few harmonic sources; a salient property of MRI acoustic noise. Both of the approaches must compensate the frequency dependent damping of the measurement chain.

3:15

1pSP7. Enhancing angular resolution by coherent beamforming of full non-uniformly spaced linear hydrophone array. Delin Wang and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., 302 Stearns Ctr., Rm. 312, Boston, MA 02115, wang.del@husky.neu.edu)

Uniformly-spaced apertures or subapertures of discrete linear receiver arrays are often used in remote sensing to increase signal-to-noise ratio (SNR) by coherent beamforming which reduces noise coming from directions outside the signal beam. To avoid grating lobes in real spatial directions, the uniformly-spaced array inter-element spacing d determines the maximum or cut-off frequency $f<c/2d$ of signals suitable for beamforming with the array, where c is the wave speed. Here we show that a non-uniformly spaced array, for instance formed by combining multiple uniformly-spaced subapertures of a nested linear array, can significantly improve the angular resolution while simultaneously avoiding dominant grating lobes in real angular space, even for signals with frequencies beyond the cut-off. The array gain, beamwidth, and maximum grating lobe height are calculated for the ONR Five Octave Research Array (FORA) for various combinations of its uniformly-spaced subapertures, including the full non-uniformly spaced array. Examples are provided of angular resolution enhancement with resulting non-uniformly spaced array or subarray in both active and passive ocean acoustic waveguide remote sensing over that of the uniformly spaced counterpart, including continental-shelf scale imagery of fish populations, and marine mammal vocalization SNR enhancement which improves detection and bearing-time estimation for passive localization.

3:35

1pSP8. Underwater target detection based on fourth-order cumulants beamforming. Xiukun Li, Hongjian Jia, and Mei Yang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 1301, Shuisheng Bldg., No.145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, jiahongjian_hrbeu@outlook.com)

Obtaining the target echo data of high signal to reverberation ratio (SRR) or signal to noise ratio (SNR) in the complex ocean environment is a key problem to improve the detection capability of underwater targets. In this paper, combined with the expansion characteristics of fourth-order cumulants to the array aperture and the structure of the smallest redundant array, a robust and high-resolution fourth-order cumulants beamforming method based on the smallest redundant array is presented. The target direction-of-arrival estimation (DOA) is achieved by forming a narrow space beam, while the reverberation interference is reduced. For the broadband detection system, combined with fractional Fourier transform (FRFT), the fourth-order cumulants beamforming method is extended to the DOA estimation of LFM signal. According to the signal characteristics difference between the target echo and the reverberation, the reverberation is filtered out in the FRFT domain, which achieves the secondary suppression of reverberation interference in the spatial domain and the transform domain, and further improves the SRR of the target echo signal. The validity of the research method is verified by the processing results of the active sonar target detection experiment.

3:50

1pSP9. Research on distributed beam forming technology in underwater communication. Hai X. Sun, Qi Jie, and liao C. Qiang (Communications Eng., Xiamen Univ. School of Information Sci. and Eng., Xiamen, FuJian 361005, China, hxsun@xmu.edu.cn)

In underwater collaborative network communication, the use of distributed beam forming technology can greatly improve the communication speed and reliability of the network, thus improving the performance of the system. Currently, a vector code based on the feedback information from the destination node to the relay node is used to design the vector code base of the beam, which is widely used in wireless communication. Because of the complex multi-diameter effect of the sound channel and the severity of the variability, the probability of the feedback error is very high. This can be fatal to the system of designing the beam forming technology through feedback information. This paper analyzes the method of feedback error of distributed beam forming system, and gives the approximate expression of feedback error probability and system performance. This paper presents a scheme to improve the feedback information by using the numerical search to improve the performance of feedback information. Through the simulation experiment analysis, the proposed scheme can obviously improve the data rate of the system.

4:05

1pSP10. Design of an unmanned aerial vehicle based on acoustic navigation algorithm. Meng Y. Gong, Hu P. Xu (Logistics Eng. College, Wuhan Univ. of Technol., Yujiayou Campus, Wuhan, Hubei 430063, China, 476793382@qq.com), and Yu H. Hu (Univ. of Wisconsin-Madison, Madison, American Samoa)

Autonomous unmanned aerial vehicles (UAV) navigate using Global Positioning System (GPS). However, in places where GPS signals are interfered, or blocked (e.g., tunnel, downtown with high-rise buildings), auxiliary navigation systems are desirable. In this work, an acoustic based near field ($<$ 1km) UAV guidance system is proposed. This system utilizes acoustic phase array mounted on UAV to provide direction of arrival estimates of narrow band acoustic sources at known locations. As such, the positions of UAV may be easily tracked. Through sensor fusion, this acoustic navigation system may also work together with GPS to provide even more accurate estimates.

Session 1pUW

Underwater Acoustics: Underwater Acoustic Propagation: Models, Methods, and Statistics

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Anthony L. Bonomo, Cochair
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Contributed Papers

1:00

1pUW1. Estimation of the probability density function of transmission loss in the ocean using area statistics. Brandon Patterson and David R. Dowling (Univ. of Michigan, 1231 Beal Ave., Rm. 2016, Ann Arbor, MI 48109, awesome@umich.edu)

Predictions of acoustic transmission loss (TL) in the ocean and its uncertainty are useful for a variety of ocean and naval engineering applications. Previously, an ad-hoc technique, area statistics (AS), was developed as a computationally efficient method to estimate the probability density function (PDF) of TL in any uncertain ocean environment from a single range-depth TL-field calculation in that environment. Here, the performance of AS is reported for ten ocean environments having uncertain sound speed profile, bathymetry, and seabed properties. In each environment, a parabolic-equation-based prediction of TL from a point source was generated as a function of range and depth at frequencies of 100, 200, and 300 Hz using the most probable values of the uncertain environmental parameters. From these calculations, AS was used to estimate the PDF of TL at more than 3000 locations within the ten environments. These estimated PDFs are then quantitatively compared, via an L1-error norm, to PDFs generated from traditional Monte Carlo techniques for the same locations and frequencies. Current results show that the L1-error of the AS PDF is less than 0.5, indicating 75% or greater overlap between the AS and Monte Carlo PDFs, at 90% of the test locations. [Sponsored by ONR.]

1:15

1pUW2. Statistics of bottom-diffracted surface-reflected arrivals in ocean acoustics. Ralph A. Stephen (RASCON Assoc. LLC, 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), S. T. Bolmer (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Bottom-diffracted surface-reflected (BDSR) arrivals are a ubiquitous feature in long-range ocean acoustic propagation and are not predicted by existing forward models based on available bathymetric and bottom properties data. In a research cruise in the North Pacific in 2013 over 40 distinct bottom diffractor locations were identified within a 25 km radius survey (at least one diffractor location every 50 square kilometers). The BDSRs can be characterized in terms of: (a) grazing angle of the incident field, (b) in-plane or out-of-plane diffractors, (c) frequency (transmissions were made from 77.5 to 310 Hz), (d) receiver type (vertical or horizontal seismometer, hydrophone, etc.), (e) receiver location, (f) signal strength relative to direct and water multiple paths, and (g) location relative to bathymetric features. In this talk, we present the statistics of the BDSRs in terms of these criteria. This information will inform efforts to predict and exploit BDSRs and will be valuable in planning future cruises to identify the geologic features responsible for them. [Work supported by ONR.]

1:30

1pUW3. Analysis and comparison of spatial correlation and intensity distribution statistics. Chad M. Smith and Daniel C. Brown (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu)

Active sonar processing systems operating in shallow-water littoral environments are often plagued with excessive false alarms, generally referred to as sonar clutter. At times the effects of these clutter events can be mitigated through repeated measurements and/or careful sonar operator examination; however, this process can be time consuming and even experienced sonar operators may have difficulty identifying targets in cluttered environments. Additionally, increasing interest in autonomous vehicle platforms and the push for real-time decision and performance management introduces further requirements on sonar systems. This work provides an empirical analysis of the potential in using spatial correlation of the backscattered wavefront as a classification metric, and compares this metric to a commonly used probability density function (PDF) processing method. The shape of the PDF of the backscattered acoustic intensity is often used in sonar processing systems such as constant false alarm rate (CFAR) detectors. These PDFs generally provide an estimate of the Rayleigh-like (random phase scattering) or non-Rayleigh-like (generally finite bright scatterers) nature of temporal variations in acoustic returns. This talk will outline information that may be available about clutter events based on measurements of spatial correlation. [Work supported by Office of Naval Research.]

1:45

1pUW4. Sensitivity of acoustic propagation modeling to variations in ocean environmental conditions. Tetyana Margolina, John E. Joseph (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, tmargoli@nps.edu), and Mary Jordan (Meteorol., Naval Postgrad. School, Monterey, CA)

A parabolic equation acoustic model has been used to estimate transmission loss for a mid-frequency sound source to assist in marine mammal behavioral response studies. The sound propagation is modeled for a range-dependent environment using the Navy Digital Bathymetric Data Base and Bottom Sediment Type database, NOAA Global Ocean Sediment Thickness Dataset, and the High-Resolution Global Sea Surface Wind Speed monthly climatology. The sound speed along the propagation path is modeled using two ocean temperature/salinity fields of different spatial and temporal resolution: monthly 0.25 deg outputs of the Generalized Digital Environmental Model (GDEM), and the daily 1/12 deg outputs of the regional Hybrid Coordinate Ocean Model (HYCOM). The modeling was done for two operational areas with strikingly different ocean environments, the Southern California Bight and the Gulf Stream region off Cape Hatteras. The latter represents a much more dynamic ocean environment with profound temperature fronts, highly variable meso- and submesoscale eddy activity, and sharp bathymetry gradients along the shelf break. Received sound levels

measured on tagged marine mammals, and synchronized sound source/sound receiver tracklines are used to estimate uncertainty of the acoustic modeling results, and the non-linear sensitivity of the model to wide-range variations in the ocean environmental conditions.

2:00

1pUW5. Geoacoustic sediment model discrimination using acoustic color comparison. Anthony L. Bonomo and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Many competing geoacoustic models exist to represent sandy sediments. These models can be fluid, elastic, or poroelastic, depending on the number and types of waves the sediment is assumed to support. In this talk, a fully scattered field finite element approach is utilized to construct acoustic color templates for a buried spherical aluminum shell and a spherical bubble. The simulated results are then used to assess the ability of target strength measurements to discriminate between the predictions of the competing sediment models. Such discrimination may help determine the physical validity of a given geoacoustic model and aid in model selection. Five geoacoustic sediment models are considered: a simple fluid, constant-Q fluid model, the effective density fluid model of Williams, the viscous grain shearing model of Buckingham, the Biot-Stoll poroelastic model, and the extended Biot model of Chotiros. [Work supported by ONR, Ocean Acoustics.]

2:15

1pUW6. Finite impulse response of an ocean acoustic waveguide by compressed sensing. Richard S. Keiffer (Acoust. Div., Naval Res. Lab., Bldg. 1005, Rm. C-2, Stennis Space Ctr., MS 39529, richard.keiffer@nrlssc.navy.mil)

The usual approach to calculating the time domain acoustic response of an ocean waveguide is to Fourier synthesize it from a regularly sampled collection of frequency domain solutions. Depending on the details of the waveguide and the excitation signal, the resulting time series may be very sparse with significant time intervals of relative quiet between energetic multipath arrivals. This observed sparsity in the time domain suggests a computational advantage to synthesizing the time series by solving a convex optimization problem involving a much smaller set of solutions computed at random frequencies. The efficiency of the compressed sensing approach will depend on the sparsity of the time domain solution; the greater the sparsity in the time domain signal the greater the expected advantage of the compressed sensing approach. In this talk we examine the compressed sensing method of time series synthesis for a sequence of waveguides that range from extremely simple and ideal waveguides where the sparsity is expected to be great to the more realistic and complicated where the sparsity is expected to be reduced. [This work was supported by the U.S. Office of Naval Research.]

2:30

1pUW7. Discrete element method simulations for acoustic propagation through idealized seafloor sediments. Joseph Calantoni, Adam Seyfarth, Pedro M. Jordan, and Richard S. Keiffer (U.S. Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529, joe.calantoni@nrlssc.navy.mil)

We performed discrete element method (DEM) simulations for the propagation of pressure and shear waves in idealized seafloor waveguides. Preliminary simulations used identical spheres configured in a random loose pack. DEM waveguides were essentially one-dimension with a 10 by 10 grain diameter cross section and varied in length from 20,000 to 100,000 grain diameters for grains ranging from 0.2 to 0.5 mm in diameter with the material properties of quartz. Rough walls were placed at the ends of the waveguide and periodic boundaries were used in the lateral direction. Pressure and shear wave propagation was simulated through normal and tangential sinusoidal oscillation of the boundary wall at frequencies ranging from 100 to 1,000 Hz. Sensitivity of propagation speed and attenuation to the specification of the material properties of the grains was investigated. For example, our simulations demonstrate the well-known dependence of pressure wave propagation speed on grain stiffness. Discussion will focus on creating links between simulation results and continuum models for acoustic propagation in seafloor sediments.

2:45–3:00 Break

3:00

1pUW8. Measurements of the acoustic-field autoproducts. Jessica E. Lipa (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133), Brian M. Worthmann (Appl. Phys., Univ. of Michigan, Ann Arbor, MI), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI, drd@umich.edu)

The frequency-difference and frequency-sum autoproducts are quadratic products of acoustic fields at different frequencies extracted from a frequency-domain acoustic field with nonzero bandwidth. In well-ensonified regions of ray-path environments, the autoproducts may—in theory—mimic genuine acoustic fields at frequencies below and above the original field's bandwidth (Worthmann and Dowling, 2017, JASA **141**, 4579-4590). For example, in a Lloyd's mirror environment, the autoproducts successfully mimic genuine out-of-band acoustic fields except near the surface or the source. In this presentation, results from a water tank experiment are shown that confirm predictions from theory. A 40- to 110-kHz pulse signal was broadcast from a source 20 cm below the surface to a hydrophone positioned at three different ranges between 20 and 50 cm and at depths from 0 to 40 cm. The measured frequency-difference autoproduct fields have cross correlations of greater than 90% with genuine acoustic fields between 0 and 60 kHz. Similarly, the measured frequency-sum autoproduct fields have cross correlations of greater than 90% with genuine acoustic fields between 110 and 190 kHz. Furthermore, the experimentally measured autoproducts correlate well with theoretically predicted autoproducts. Thus, these results serve to confirm the theory. [Sponsored by ONR and NSF.]

3:15

1pUW9. Autoproducts in refractive waveguides near shadow zones. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The frequency-difference and frequency-sum autoproducts are quadratic products of nonzero-bandwidth frequency-domain acoustic fields at different frequencies but the same spatial location. The autoproducts have been shown to mimic genuine lower- or higher-frequency, out-of-band, acoustic fields within some limitations (Worthmann and Dowling, 2017, JASA **141**, 4579–4590). Previous analysis of the autoproducts considered only spatial regions of the environment that were well-ensonified by the in-band acoustic field, which itself is well-described by the ray-approximation. In this presentation, the necessity of this well-ensonified restriction is analyzed by comparing autoproduct fields with their out-of-band analogues in and near acoustic shadow zones created by refraction. The results presented are derived from exact or asymptotic solutions of the Helmholtz equation. The propagation environment chosen for this study is a horizontal, range-independent, stratified, unbounded, truncated-n^2-parabolic waveguide, which creates acoustic shadow zones in some regions of the acoustic field. By analyzing the autoproducts near the transition from well-ensonified region to shadow zone, the potentially detrimental effects of diffraction on the autoproducts' mimicry of out-of-band acoustic fields can be analyzed and quantified. [Sponsored by ONR and NSF.]

3:30

1pUW10. A higher-order approximation to a hybrid parabolic equation approach for density discontinuities. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu)

Parabolic equation models based on the highly efficient and stable split-step Fourier (SSF) algorithm have been known to suffer from accumulating phase errors in range when bottom interactions are prevalent. The primary cause has been identified as due to the treatment of the density discontinuity at the water/sediment interface, whereby a smoothing function is introduced to model the step function. The scale of this smoothing function is proportional to wavelength, and can distort the environmental representation at low frequencies. Yevick and Thomson (1997) introduced a hybrid approach to the problem whereby the density terms were treated separately from the

SSF approach via a finite-difference approximation. While this approach was shown to significantly reduce the aforementioned phase errors, the solutions were found to be highly sensitive to the choice of depth mesh size, making general usage of this approach less optimal. The finite difference approach of Yevick and Thomson was based on a 2^{nd} order approximation of the differential operators. In this work, higher order approximations are applied in an attempt to stabilize the finite difference approximations. Solutions are presented for a simple shallow water waveguide utilizing the various approaches previously investigated as well as the higher order approximations. The results are compared with respect to both solution accuracy as well as numerical stability.

3:45

1pUW11. Exploration of acoustic spiral waves in an underwater environment. Grant Eastland (Test and Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The work to be presented is regarding a research project to investigate the viability and possible application of spiral or helical acoustic waves for use in underwater environments. The investigation encompasses studies into the properties and generation of spiral or helical acoustic waves. Spiral waves can be generated by precisely controlling the phase of the wave, which could have different physical properties due to what is known as "topological charge." This topological charge could be adjusted in some advantageous way, depending upon the desired application. In addition, reception of spiral or helical waves was investigated. The goal of the research is a proof of concept and presentation of the results of generation and reception of spiral acoustic waves. Spiral waves have been investigated for acoustic navigation and may lend itself well for use in other applications.

4:00

1pUW12. Using the argument of the Airy function as an analysis tool in ocean acoustics and oceanography. Stanley A. Chin-Bing (Phys. Dept., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chinbing@att.net) and Josie P. Fabre (Acoust. Div. Code 7180, Naval Res. Lab., Stennis Space Ctr., MS)

By assuming the index of refraction is linear between layers, it is possible to convert the depth dependent acoustic Helmholtz equation into an Airy equation. Decades ago this allowed the ocean acoustic pressure to be quickly calculated using tables of precalculated Airy functions with interpolation. High speed computers and the accurate parabolic equation model have rendered the Airy equation approach as a curiosity of history. However, the method can still be useful as an assessment tool. This has been demonstrated by using it to select similar ocean environmental parameters that have the same acoustic propagation characteristics. In this presentation, we apply the Airy equation method to evaluate the buoyancy frequency of similar ocean environments that support internal waves. A connection is made between the ocean acoustic propagation and the oceanography of the similar regions.

4:15

1pUW13. A fast calculation method of travel speed of pulse peak in convergence zone. Yun Ren, Renhe Zhang, Jun Wang, and Yonggang Guo (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renyun@mail.ioa.ac.cn)

A long-range sound propagation experiment was conducted in the West Pacific Ocean in summer 2013. The signals received by a towed array indicate that the travel speed of pulse peak (TSPP) in the convergence zones is stable. Therefore, an equivalent sound speed can be used at all ranges in the convergence zones. A fast calculation method based on the beam-displacement ray-mode (BDRM) theory and convergence zone theory is proposed to calculate this equivalent sound speed. The computation speed of this proposed method is over 1000 times faster than that of the conventional calculation method based on the normal mode theory, with the computation error less than 0.4% compared with the experimental result. Also, the effect of frequency and sound speed profile on the TSPP is studied with the conventional and fast calculation methods, showing that the TSPP is almost independent of the frequency and sound speed profile in the ocean surface layer.

4:30

1pUW14. Analysis of sound propagation from the transitional area to deep water. Jixing Qin, Renhe Zhang, and Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd. Haidian District, Beijing 100190, China, qjx@mail.ioa.ac.cn)

Underwater acoustic propagation over the continental shelf and slope is complicated and is also an important issue. Motivated by a phenomenon in an experiment conducted in the South China Sea indicating that the energy of the received signal around the sound channel axis is significantly stronger than that at shallower depths, we study sound propagation from the transitional area to deep water. Numerical simulations with different source depths are first performed, from which we reach the following conclusions. When the source is placed near the sea surface, acoustic wave will be strongly attenuated by bottom losses in a range-independent deep-water environment, whereas it can propagate to a very long distance because of the continental slope. When the source is mounted on the slope bottom in shallow-water area, acoustic energy will be trapped near the sound channel axis, and it converges more evidently than the case where the source is located near the surface. Then, simulations with different source ranges are performed. By comparing the relative energy level in the vertical direction between the numerical and experimental results, the range of the unknown air-gun source is roughly determined. [Work supported by the National Natural Science Foundation of China under Grant Nos. 11434012 and 41561144006.]

Session 1eIDa

Interdisciplinary: Special Presentation on The Clarinet in Early New Orleans Jazz

David Woolworth, Chair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

Chair's Introduction—5:00 p.m.

Invited Paper

5:05

1eIDa1. The Clarinet in Early New Orleans Jazz: Style, Tone and Function. Michael White (Xavier Univ. of Louisiana, 1 Drexel Dr., New Orleans, LA 70125)

Very different from its sound in classical music, swing, modern jazz or other genres, the authentic New Orleans Jazz clarinet tradition remains among the most distinctive instrumental styles in American music. This program will focus on sound and tonal variety of the clarinet in early jazz in its role as an accompanying or featured instrument in dance ensembles and its high wailing voice in brass bands and jazz funerals. There will be a discussion of the instrument's distinguishing characteristics and tonal possibilities. The high level of diversity of sound among early jazz clarinetists will also be examined as affected by equipment, training and physical characteristics of individual players. Live musical examples will be provided by the Dr. Michael White Quartet. Dr. Michael White is a leading figure in traditional New Orleans jazz and one of only a few to creatively carry on the rich clarinet sound and style of that city. For over two decades he has been the main consultant for traditional jazz for the New Orleans Jazz & Heritage Festival. A relative of several first generation jazz musicians, he has distinguished himself as a jazz historian, writer, producer, and composer. In 2015 he received the Jazz Hero Award from the Jazz Journalists Association. Dr. White earned his PhD in Spanish at Tulane University and currently holds the Charles Keller Endowed Chair in the Humanities at Xavier University of Louisiana.

Payment of a separate registration fee is required to attend this session.

MONDAY EVENING, 4 DECEMBER 2017 SALON D, 7:00 P.M. TO 9:00 P.M.

Session 1eIDb

Interdisciplinary: Tutorial Lecture on Infrasound Phenomenology, Propagation, and Detection

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

Chair's Introduction—7:00

Invited Paper

7:05

1eID1b. Infrasound phenomenology, propagation, and detection. Roger M. Waxler (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu)

Infrasound is generally understood to refer to acoustic disturbances in the atmosphere in frequency bands below the threshold of human hearing, but above the frequencies at which internal gravity waves propagate. The infrasonic band is nominally taken to be 0.05 Hz up to about 20 Hz. Infrasonic signals tend to be generated by large, violent events and propagate efficiently in sound ducts formed by wind jets and temperature gradients in the middle and upper atmosphere. Infrasonic signals can be detected at very large distances from the source, sometimes even globally. This tutorial will present an overview of the generation of infrasound by both natural and anthropomorphic sources and of the subsequent propagation and detection of infrasonic signals. Two particular sources will be discussed in detail: the signal generated by a large explosion and the so-called microbarom signal generated by colliding ocean waves. Signal propagation through the atmosphere will then be discussed. Available open source infrasound propagation packages and their use will be introduced, the significant atmospheric sound ducts will be identified, and the difficulties inherent in modeling propagation through a dynamic atmosphere will be emphasized. The current state and availability of atmospheric specification will be touched upon. Finally, the use of array processing for the extraction of infrasonic signals from the pressure fluctuations inherent to a turbulent atmosphere will be discussed.

Session 2aAA

Architectural Acoustics and Musical Acoustics: Performance Spaces for Modern Music

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

Bruce Olson, Cochair
Olson Sound Design LLC, 8717 Humbolt Ave. N., Brooklyn Park, MN 55444

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Historical evolution of sound diffusion in spaces for modern music. Peter D'Antonio, Jeffrey Madison (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), and Trevor J. Cox (Acoust. Eng., Univ. of Salford, Salford, United Kingdom)

This presentation will review the historical evolution of the use of sound diffusors, from their initial use in recording control rooms to include almost all spaces for the performance, recording, and audition of music. Shortly after the invention of the reflection phase grating diffusor by Manfred Schroeder in 1973, the quadratic residue diffusor was installed in Michael Fowler Hall, New Zealand, Underground Sound recording studios in Largo, MD, and in the Oak Ridge Boy's Acorn Studio in Henderson, TN. The initial acceptance of these new surfaces led to installation in hundreds of recording control rooms and live rooms, home theaters, stages, auditoria, and worship spaces. The design of sound diffusors was expanded to use optimization algorithms created by Cox and D'Antonio to include decorative shapes to complement the original phase grating surfaces, thus opening their use in all architectural acoustic spaces. Following a brief review of the design theory options, many applications over three decades, in a wide range of venues will be presented. A speculation on what the future holds, based on evolving diffusive designs and advanced manufacturing methods, will also be presented.

8:25

2aAA2. Taming reverberation in an outdoor amphitheatre—The Ford. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The historic Ford Amphitheatre in Hollywood, CA, recently completed an overall renovation and expansion. The centerpiece is the 1,200-seat outdoor amphitheater, which features a dramatic arroyo backdrop. The side walls, rear wall, towers, and house flooring were built of concrete in 1931 after a brush fire destroyed the original 1920 wood structure. Several sound-absorptive treatments had been applied to the walls over the years previous to our involvement, but excessive reverberation and some anomalous reflections remained. The new design features an expanded "sound wall" that helps to mitigate highway noise while providing optimal lighting and control positions, and offering an opportunity to improve the acoustical treatment scheme. The result is a unique installation of decorative perforated-metal panels that tame the reverberation, especially in the low frequencies. Assorted design challenges, apparent arroyo hillside contributions, and the resultant reverberation will be discussed.

8:45

2aAA3. Active Acoustics at the Appel Room, Jazz at Lincoln Center. Tom Wetmore (Columbia Univ., New York City, NY) and Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

The Appel Room (formerly Allen Room) is a 483-seat venue used primarily for jazz performances in the Jazz at Lincoln Center complex in New York City. In 2013 the Constellation active acoustic system by Meyer Sound was incorporated into the room's architecture. The system allows a wide range of acoustic conditions to be set for any performance. In the first four years of operation, a variety of solo and ensemble jazz artists have performed in this venue. This paper will describe the room, the system, and the results over the first four years of operation with a variety of musicians.

9:05

2aAA4. Amplified rehearsal and performance spaces on a community college campus. Sam Ortallono (Visual Performing Arts, Lee College, 711 W. Texas Ave., Baytown, TX 77522, sortallono@lee.edu)

Amplified rehearsal and performance spaces on a community college campus. Lee college in Baytown Texas has several rehearsal and performance spaces available to presenters and performers. Many of these spaces have access to sound reinforcement which interacts with the structure. In this presentation, we compare and contrast the amplified systems and spaces, including variable acoustic treatment. Each space has a dedicated purpose that determines the equipment.

9:25

2aAA5. It's All About That Bass, Design of a Showroom for Music, Comedy, and Theater. Bruce C. Olson (Olson Sound Design LLC, 8717 Humboldt Ave. North, Brooklyn Park, MN 55444, Bruce.Olson@afmg.eu) and Ana M. Jaramillo (AFMG Services North America LLC, Brooklyn Park, MN)

An 1800 seat casino showroom was designed to provide "state-of-the-art" acoustics for superstar country and pop performers, headlining comedians, and big production Broadway musicals. Challenges included a very wide stage, a curved back wall and a client requirement for hidden loudspeakers. This paper will walk you through how these challenges were met while maintaining the high-class look and feel of the room. We will also show the evolution of the sound system design from the initial system through successive upgrades.

9:45–10:00 Break

10:00

2aAA6. Design of halls for the enjoyment of American music. Richard Talaske (TALASKE | Sound Thinking, 1033 South Boulevard, Oak Park, IL 60302, rick@talaske.com)

Designing for contemporary music performance is more involved than achieving the accepted mid-frequency reverberation time within the performance space. While accomplishing the correct degree of reverberance is essential, this alone is not a sufficient condition for creating an acoustic environment suitable for the enjoyment of jazz, blues, folk, rock, computer, world, or other non-symphonic or non-choral genres of music. Control of low-pitched sound, achieving moderate running liveliness, and creating a microphone-friendly acoustic environment are also essential design goals. As usual, the acoustic goals should be accomplished while developing a welcoming room design which offers intimacy between performers and patrons and a communal audience setting. These acoustical and societal goals are discussed in relation to completed projects such as the Ellis Marsalis Center for Music in NOLA, The Sidney Harman Center in DC, and the Old Town School of Folk Music in Chicago.

10:20

2aAA7. The Great Outdoors— The unique challenges imposed by the outdoor environment for both sound reinforcement, as well as acoustical enhancement, of acoustical and amplified music. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Jonathan Laney (d&b audiotechnik, Ashville, NC)

Outdoor performance venues epitomize the definition of a multi-purpose performing arts facility. However, they differ in many ways from their indoor counterparts. They can be larger than many arenas, but lack the architecture that both provides the aural cues of a large volume, and in addition, provides insulation from intrusive noise. The programming is often more diverse. One day the venue could host a three or four act amplified concert—the following day a symphony orchestra, or a week long music festival. Weather is also an enormous unknown factor. We will discuss the changing nature of both the programming, as well as listeners expectations, at these venues. We will also discuss the technologies employed at several outdoor venues that enable them to meet the expectations of the musicians as well as the audience.

10:40

2aAA8. A 240-Seat Recital Space for Amplified Performance. Damian Doria (Stages Consultants LLC, 75 Feather Ln., Guilford, CT 06437-4907, damianjdoria@gmail.com) and Dave Kotch (Criterion Acoust., Jersey City, NJ)

A new 240-seat recital space completed in 2017 for a Canadian University with a Bachelor of Music program teaching jazz, rock, pop, metal, blues, soul, county, electronica, and hip hop. The architectural design brief required a room that could accommodate the diverse programming of student recitals and other music sessions in a modern an aesthetically satisfying interior design. The concept for the room was quickly adopted by the Owner and Design Team, but included several room acoustics challenges. This paper will discuss the acoustical accommodation of activities in the recital space, along with integration of the house sound system, and a chronicle of the design, value-engineering, construction, and final commissioning process.

11:00

2aAA9. Ninety never sounded so good: Guiding an historical auditorium into the 21st century. Brandon Cudequest and David A. Conant (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, bcudequest@gmail.com)

The Sacramento War Memorial Auditorium opened in 1927 with a screening of the silent film, "Old Ironsides." Over its ninety-year career, the 3500-seat auditorium has hosted everything from Shriners conventions to Shinedown concerts, Ringling Brothers to Rolling Stones, and many more. In its current state, the room is excessively reverberant for amplified music. Heating, ventilation, and air conditioning systems were upgraded during the 1990s, albeit without apparent acoustical input. Street traffic noise is clearly transmitted via acoustically weak windows and stage house doors. McKay Conant Hoover is currently engaged in an acoustical, audiovisual, and

theatrical renovation of this historic venue. Importantly, the Auditorium must temporarily accommodate Sacramento's touring Broadway shows, Philharmonic, choral, and ballet performances while their main venue, The Community Center Theater, undergoes a several year-long renovation. This paper will discuss acoustical challenges faced, current findings, and how factors such as ill-conceived pit lifts, historical finishes, limited storage, and lead paint influenced our recommendations.

11:20

2aAA10. Worship space acoustics and architecture for contemporary services with modern music. Robert C. Coffeen (None, R. C. Coffeen Consultant in Acoust., Lawrence, KS 66047, bob@rccoffeen.net)

In relatively recent time, some religious worship facilities are presenting a contemporary worship service with music that can be considered modern and different from the more traditional music produced by an organ, piano, groups that employ instruments as used by a symphony orchestra, and by a choir. Modern music for a contemporary service is typically produced by a band using electric keyboards, electric and acoustic guitars, drums, and singing by a solo vocalist or by singing groups with all of the music electro-acoustically amplified. Thus, halls, auditoriums, and sanctuaries for contemporary worship services must generally be less reverberant than a facility for traditional music, and the sound reinforcement system must properly handle relatively high level music and vocals. It also seems that architecture for worship facilities where contemporary services will be held has changed from more traditional architecture to more informal designs with exposed ceiling structure, movable chairs, etc. This also seems to produce HVAC systems with higher noise due to exposed supply air ducts, short return air paths, and roof-top air handlers. Examples of and acoustical data for worship spaces with contemporary services will be presented.

TUESDAY MORNING, 5 DECEMBER 2017 — SALON E, 8:35 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics, Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography: In Memory of George Ioup: Acoustics in the Gulf of Mexico I

Natalia Sidorovskaia, Cochair
Physics, UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210

David K. Mellinger, Cochair
Coop. Inst. for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Chair's Introduction—8:35

Invited Papers

8:40

2aAB1. George Ioup's contribution to the Gulf of Mexico acoustic research: paving the path into the future. Natalia Sidorovskaia (Phys., UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu) and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

By the end of the 1990s, researchers and regulators recognized the need for understanding how anthropogenic activities impact cetacean's populations in the Gulf of Mexico. In 2000, George Ioup was one of the founders of the Littoral Acoustic Demonstration Center (LADC), a consortium of Gulf Coast scientists, with the long-term goal of studying the anthropogenic soundscapes of the Gulf of Mexico and their impact on marine mammals. In 2001, LADC was the first team to collect long-term acoustic data rich in sperm whale phonations from bottom-moored autonomous buoys, technology developed by NAVOCEANO and adapted for LADC needs. The first step in establishing a baseline database was taken. At the same time, George sparked the interest of bioacousticians with ideas on how to employ the differences in sperm whale phonations to identify individuals similar to how humans recognize voices. LADC endeavors continued through designing acoustic surveys to characterize the soundscapes of seismic exploration arrays and being the first team to record beaked whales in the Gulf of Mexico in 2007. Recent advances in understanding marine mammal habitats, studying the oil spill impact on these animals, and introducing new acoustic technologies would not be possible without the seminal work of George Ioup.

9:00

2aAB2. A comparison of automated and manual techniques for acoustically identifying individual sperm whales with changing aspect. Christopher Tiemann (R2Sonic LLC, 5307 Industrial Oaks Blvd., Ste. 120, Austin, TX 78735, chris@r2sonic.com)

As shown by his extensive history of research in the Gulf of Mexico, Dr. George Ioup had a fascination with the marine mammals that lived there. In particular, he questioned whether individual whales of a given species could identify each other acoustically, and by extension, whether humans could do the same. George was an early proponent in the community of advancing automated methods for discriminating individual odontocetes by analyzing the clicks they make, a problem that grows considerably more difficult when multiple animals are clicking simultaneously. I had the good fortune of being able to explore this subject with George in an independent research project that may not have been widely reported but was important and enjoyable for us nonetheless. George realized quickly that conventional methods for automated grouping of clicks from the same click train were failing because there was no guarantee of click structure consistency as an animal changed aspect relative to a receiver. An automated method for grouping sperm whale clicks based on a cross correlation method originally applied to dolphin clicks provided the needed flexibility to follow an evolving click shape as shown by the close match between George's automated results and my manually generated groupings.

9:20

2aAB3. Modeling as a complementary tool to acoustic data for understanding the impact of environmental disasters on marine mammals. Azmy S. Ackleh, Ross Chiquet, Tingting Tang, Amy Veprauskas (Mathematics, Univ. of Louisiana at Lafayette, P.O. Box 41010, Lafayette, LA 70504, ackleh@louisiana.edu), Hal Cawell (Univ. of Amsterdam, Amsterdam, Netherlands), Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), and Baoling Ma (Millersville Univ., Millersville, PA)

This study is focused on how environmental disasters, such as the Deepwater Horizon oil rig explosion in 2010, affect the dynamics of marine mammal populations, particularly sperm whales and beaked whales, in the Northern Gulf of Mexico. We briefly describe how modeling techniques are used to estimate densities of marine mammals using passive acoustic data. We then develop a matrix model to examine the possible long-term effects of a disaster. We consider cases in which the effects of a disturbance result in reductions in either survival (lethal impacts) or fecundity (sublethal impacts). This model, combined with demographic stochasticity, allows us to study the long-term recovery process following an environmental disaster. In particular, recovery probabilities and recovery times of the population are computed, and formulas are derived to compute the sensitivity of the recovery time to changes in properties of the population or the environmental disturbance. We then extend the modeling methodology to consider how marine mammals may be affected by the response of their prey population to a disturbance. Our analysis highlights the difficulty of projecting impacts and recovery in the absence of detailed demographic data, and the value of population models in exploring scenarios and identifying important processes and general relationships. [This research was made possible in part by a grant from The Gulf of Mexico Research Initiative.]

9:40

2aAB4. Passive acoustic monitoring in the Northern Gulf of Mexico using ocean gliders. David K. Mellinger, Sharon L. Nieukirk, Sara L. Heimlich, Selene Fregosi (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Elizabeth T. Küsel, Martin Siderius (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., Portland, OR), and Natalia Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

Although George Ioup did not use ocean gliders for passive acoustic monitoring, he recognized their value as platforms for PAM and encouraged others to use them. They function well as PAM platforms because (1) they move slowly, minimizing flow noise; (2) they have no propeller or continuously running machinery, minimizing motor noise; (3) they collect acoustic data nearly continuously; (4) they traverse the upper water column every few hours, measuring temperature and salinity as needed for calculating sound speed profiles and enabling accurate modeling of long-range acoustic propagation; (5) they can cover hundreds to thousands of kilometers in distance during a deployment, enabling them to monitor a large area and/or repeatedly monitor a smaller area; and (6) some models can dive to 1000 m, the depth at which some deep-diving cetaceans—sperm and beaked whales, frequent targets of PAM operations—forage and vocalize. Two models of gliders equipped with passive acoustic recording systems were deployed in the Northern Gulf of Mexico in the summers of 2015 and 2017 to study cetacean occurrence and behavior. Here, we summarize the virtues and hazards of glider PAM, and describe acoustic detection and classification of cetaceans in these recordings. [Research supported by GoMRI.]

10:00

2aAB5. Echolocation for restoration: Odontocete monitoring in the Gulf of Mexico. Kaitlin E. Frasier, Rebecca Cohen, Jennifer S. Trickey, Sean M. Wiggins, Alba Solsona Berga (Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, kfrasier@ucsd.edu), Melissa Soldevilla, Lance Garrison (Protected Resources Div., NOAA SEFSC, Miami, FL), Simone Baumann-Pickering, and John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

In the late 1990s, George E. Ioup began studying echolocation clicks as a means of understanding marine mammals in the Gulf of Mexico (GOM). He also led one of the few research programs focused on pelagic species in this chronically impacted region in the years preceding the Deepwater Horizon oil spill. Today, passive acoustic monitoring (PAM) is one of the primary tools used to study the nearly 20 pelagic odontocete species found in the GOM, including sperm whales, beaked whales, dolphins, and *Kogia* species. Since 2010, PAM devices have been deployed nearly continuously in the region, driven by an urgent need to understand the long-term effects of both acute and chronic anthropogenic impacts on GOM marine mammal populations. Recent advances fueled by robust, reliable PAM technologies include the development of multi-year timeseries documenting changes in species densities across continental shelf and slope habitats, differentiating GOM odontocete species based on echolocation click properties, and leveraging long-term datasets to understand the influence of habitat variability on offshore species distributions. These capabilities are proving PAM is indispensable as an observation method for damage assessment, decision support, and restoration activities for marine mammals, especially inaccessible pelagic populations.

10:20–10:40 Break

10:40

2aAB6. Airborne electromagnetic profiling in the Gulf of Mexico. Kenneth W. Holladay (Mathematics Dept., Univ. of New Orleans, New Orleans, LA 70148, khollada@uno.edu)

First, I will give an overview of the research on Airborne Electromagnetic Profiling (AEM) by the University of New Orleans (UNO) group that included George Ioup. The standard techniques of error correction, calibration, modeling, inverse methods, interpolation, and smoothing were used to convert raw profiler data into meaningful data and useful products. As a part of a NASA EPSCoR project, the group applied layer models to AEM data to measure shallow water bathymetry and water column stratification in the Gulf of Mexico and coastal near surface geomorphology in the Barataria basin of Louisiana. In the second part of the talk, I will sketch the history of the UNO Engineering and Applied Science doctoral program and George Ioup's large role in it.

11:00

2aAB7. Lognormal bubble size distributions. Jerald W. Caruthers (29 Pecan Dr., Long Beach, MS 39560, jcaruth123@gmail.com)

In the study of oceanic bubbles based on empirical acoustics, the distributions among the sizes are typically represented by power laws with negative slopes as convenient descriptors of data in log/log format. However, power laws do not address the fact that as bubble sizes approach zero their numbers must approach zero. Multiple power laws, including a horizontal segment preceded by a positive power-law segment and followed by the expected negative power-law segment, have been used to recognize and mitigate this problem. Although not universally adopted as yet, several acoustic researchers have suggested that at least some oceanic bubble distributions are more appropriately represented by lognormal distributions. In 1941, A. N. Kolmogorov used the 1922 work of L. R. Richardson concerning a stochastic downward cascade of random sizes of turbulent vortices that asymptotically result in lognormal distributions of vortices. This current paper uses such a cascade of vortices to begin a downward cascade of bubbles sizes causing cascading shear forces on large bubbles that were created by breaking waves. In combination with this decreasing effect of turbulent shear on these fragmenting bubbles, the downward cascade of bubbles sizes overlaps and continues with a strengthened partial-pressure effect on ever increasing surface tension caused by their diminishing sizes. Issues associated with this approach, such as summing lognormal generators and intermittency, will be discussed.

11:20

2aAB8. In memory of George E. Ioup: Founder of the University of New Orleans physics program at the Stennis Space Center. Stanley A. Chin-Bing and Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chinbing@att.net)

In the late 1970s, two U.S. Navy organizations were formed and located in an area now known as the Stennis Space Center, Mississippi, located nearly 50 miles East of the University of New Orleans (UNO). These two organizations employed nearly 1500 scientists and technicians who needed advanced training in physics, specifically courses involving acoustics and signal processing. In 1982, Professor George E. Ioup took the initiative to have UNO develop and teach these courses on-site at the Stennis Space Center. In the following 33 years, nearly 20 different graduate level courses were developed and taught multiple times at the Stennis Space Center. It was possible to take all the necessary courses needed for the Masters and Ph.D. degrees on-site, while maintaining full-time employment. Under Professor Ioup's leadership, several dozen Navy scientists received advanced degrees from the University of New Orleans, and many more received specific training that enhanced their professional careers. This presentation will highlight the dedicated efforts and successes of George Ioup in creating the UNO physics program at the Stennis Space Center, MS.

11:40

2aAB9. Relationship between head size and biosonar transmit and receive beams in odontocetes. Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu)

The unique shape of a dolphin head, the different specie specific shape and the internal head structure suggest a very complex propagation mechanism for the biosonar signals to travel from the phonic lips into the water. Despite these factors, a circular planar aperture of the appropriate diameter can produce a transmission beam that resembles the corresponding beam of a dolphin. In the similar manner, the reception process is also very complex and not completely understood. Once again, despite the complexity involving the reception of biosonar echoes, a simple circular planar aperture of the right diameter can have a receiving beam that approximate or resemble the receiving beam of odontocetes. For both transmission and reception of broadband biosonar signals, the size of the head compared to the wavelength of the signal will determine the degree of directionality of the beams in odontocetes in a similar manner as a planar transducer. Beam pattern data from *Tursiops truncates*, *Delphinapterus leucas*, *Pseudorca crassidens*, and *Phocoena phocoena* will be used to demonstrate the relationship between head size and directionality for the transmit signal. Receiving beam pattern data from *Tursiops truncates* and *Phocoena phocoena* will be used in a similar manner as for the transmit beam.

Session 2aBA

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

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

Pierre Belanger, Cochair
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Chair's Introduction—8:25

Invited Papers

8:30

2aBA1. Parity-time synthetic phononic media. Johan Christensen (UC3M Madrid, UC3M, Madrid 28935, Spain, johan.christensen@uc3m.es)

Classical systems containing cleverly devised combinations of loss and gain elements constitute extremely rich building units that can mimic non-Hermitian properties, which conventionally are attainable in quantum mechanics only. Parity-time (PT) symmetric media, also referred to as synthetic media, have been devised in many optical systems with the ground breaking potential to create non-reciprocal structures and one-way cloaks of invisibility. Here, we demonstrate a feasible approach for the case of elasticity where the most important ingredients within synthetic materials, loss and gain, are achieved through electrically biased piezoelectric semiconductors [1]. We study first how wave attenuation and amplification can be tuned, and when combined, can give rise to a mechanical PT synthetic media with unidirectional suppressed reflectance, a feature directly applicable to evading sonar detection [2]. [1] J. Christensen, M. Willatzen, V. R. Velasco, and M.-H. Lu, *Phys. Rev. Lett.* **116**, 207601 (2016). [2] S. A. Cummer, J. Christensen, and A. Alu, Nature Rev. Mater. **1**, Article number: 16001 (2016).

8:50

2aBA2. Beyond the single-scattering assumption for analysis of diffuse ultrasonic scattering experiments. Joseph A. Turner and Nathanial Matz (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

Characterization of diffuse ultrasonic scattering is challenging because accurate models require assumptions about the degree of scattering. Most previous research has focused on the single-scattering regime in which rays are assumed to scatter only once before detection. In this presentation, scattering effects beyond single scattering are examined with a focus on polycrystalline materials. The contribution of the second scattering within the response is quantified with respect to measurement parameters and sample properties. The results show that single-scattering models are appropriate for weakly scattering materials, such as aluminum, for a wide range of experiments and grain sizes. However, stronger scattering materials are predicted to have significant components beyond single-scattering for certain measurement parameters even in the Rayleigh regime. Experimental results for a steel alloy are used to verify model predictions. The experimental work shows the domain for which the doubly-scattered response becomes significant as well as the limitation at which the double scattering model is no longer applicable. The results can be used to predict the applicable frequency range for each level of scattering for a given experimental configuration. Finally, the impact of the higher-order scattering is discussed with respect to the detection of defects within a heterogeneous medium.

9:10

2aBA3. Acoustic propagation in a fractal network of scatterers. Vincent Gibiat, Etienne Bertaud (Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr), MArie-Fraise PONGE (I2M, Bordeaux Univ., Bordeaux, France), and Xavier Jacob (Toulouse Univ., Toulouse, France)

Fractal networks, built on Cantor set, Fibonacci series or Sierpinsky sets are characteristics of a structural organization between periodic and random. On the other hand, they have been proved to be a good descriptor of irregular systems as the well known description of Brittany coasts. Such systems can be seen as irregular boundaries, as in the case of irregular coasts, while the wave equations are still usable inside theses boundaries. Along the boundaries, wave localization is possible. Another case is that where boundaries are regular while the organization of the material where waves are propagating carries the fractality. Wave propagation on such systems can be considered through the aspect of multiple scattering, and on 1D systems as well on 2D systems built on Cantor or sierpinski set, it is possible to show

the existence of localized modes. On the contrary to the previous case, localization arises not along the boundaries but inside the propagating medium. Less common organization of scatterers, for instance, those built on Fibonacci series can also be compared to periodic organizations as well as random ones. Various examples will be shown on experimental material as on theoretical or numerical examples.

9:30

2aBA4. Relationship between ultrasound scattering and acoustic impedance maps in sparse and dense random media. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org), Kazuki Tamura (Graduate School of Eng., Chiba Univ., Chiba, Japan), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Tadashi Yamaguchi (Ctr. for Frontier Medical Eng., Chiba Univ., Chiba, Japan), and Emilie Franceschini (Laboratoire de Mécanique et d'Acoustique, Aix Marseille Université, CNRS, Centrale Marseille, Marseille, France)

Quantitative modeling of ultrasound scattering from soft tissues has been used extensively to characterize soft tissues. In this approach, tissues are typically modeled as a random medium containing scatterers of specific shapes, acoustic properties, and spatial arrangements. Under plane-wave insonification and assuming weak scattering (i.e., Born approximation), the backscattered coefficient (BSC) of such a random medium is fully described by the power spectrum of its three-dimensional (3D) impedance map (ZM). A two-dimensional (2D) ZM can be obtained by scanning acoustic microscopy (SAM) applied to thin tissue sections using very high-frequency ultrasound (>100 MHz). Under isotropic assumptions, 2DZMs can predict the BSC accurately; nevertheless, in the case of dense media, where the locations of the scatterers can be correlated, some of the theoretical assumptions fail, which requires introduction of the structure-factor model (SFM). Using experimental and simulated data, this presentation will review computation of the BSC from ultrasound measurements, the working principles of SAM, the use of 3DZMs and 2DZMs to predict the BSC, SFM estimation from 2DZMs and 3DZMs, and the use of SFM for BSC computation and tissue characterization.

9:50

2aBA5. Simulation of guided waves in layered fluid/viscoelastic/poroelastic media using semi-analytical finite element method. Vu-Hieu Nguyen (Univ. of Paris-Est, 61 Ave. du général de Gaulle, Creteil 94010, France, vu-hieu.nguyen@u-pec.fr)

Understanding of the ultrasound transmission in functionally graded structures materials is of great interest in many engineering applications such as geophysics, biomedical diagnostics, aircraft, and automobile. This paper will present a computational method and its implementation procedure to study the wave propagation problem in multilayer structures made from a combination of fluid, anisotropic viscoelastic, and poroelastic materials. The poroelastic material is described by using the Biot theory. The developed approach is based on the Semi-analytical Finite Element Method (SAFE), which only requires the discretization of the cross-section of the structure. For the finite element solver, high-order spectral element has been used, showing a significant improvement of the computational efficiency compared to the use of conventional high-order elements. Numeral validation in both time and frequency domains show that the proposed approach is efficient to investigate the transient response as well as the dispersion of layered media. Some results in the context of quantitative ultrasound axial transmission techniques for assessing properties of cortical long bones will also be presented.

10:10–10:25 Break

10:25

2aBA6. Nonlinear coda wave interferometry: Characterizing damage in complex solids. Vincent Tournat (LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr)

In this talk, we report results on the nonlinear interactions of ultrasonic coda waves, in reverberating or multiple scattering mesoscopic solid media. Using the method of coda wave interferometry (CWI), we analyze the effect of mixing a coda wave with an additional lower frequency pump wave. The extracted CWI parameters, known to be highly sensitive to small geometric or elastic modifications of the tested medium, are shown to be pump-amplitude dependent and to capture finely the results of the nonlinear interactions. Although nonlinear self-action effects with coda waves have been reported in unconsolidated granular media, they are difficult to implement in cracked solids or concrete. Instead, the reported nonlinear CWI class of methods (NCWI) shows robustness, a high sensitivity, and has been applied successfully to various complex media and structures. We show through several examples on « model » media (cracked glass plates) and on concrete structures, that NCWI can be useful for the nondestructive evaluation of complex solids that are strongly scattering at usual probing frequencies. Preliminary results and prospects in nonlinear elastic properties imaging and quantitative evaluation with NCWI are discussed.

10:45

2aBA7. Propagation of ultrasonic waves in complex media: Investigation of coda wave technique and traditional polar and C-scan imaging. Nico F. Declercq, Lynda Chehami (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France, nico.declercq@me.gatech.edu), Pascal Pomarède (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France), Othmane Ez-Zahraouy, Esam Ahmed Mohammed (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, Metz, France), and Fodil Meraghni (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France)

A discussion of nondestructive techniques is presented for the investigation of complex media, with a focus on composite samples. Traditionally, one applies ultrasonic C-scans, or polar scans, which are easy to implement and to interpret. However, in many realistic cases, it is important to use more sophisticated approaches as C-scans often do not reveal any useful information. Typically, the early part of received signals is used to extract information, whereas the later part is considered either as noise or as a useless coda wave as in musical acoustics. Nevertheless, it appears that the coda part carries useful information about the medium, and therefore, it is important to explore techniques to extract that information. In addition, it turns out that the coda is very sensitive to material properties and damage as those sound waves interact longer with the material than early arrival waves. First, earlier results will be shown which compare experimental polar scans with numerical simulations, then, for the same samples, coda wave results will be presented to show the effect of damage on the composite samples. The main damage indicator is the change in relative wave velocity which is caused by the damage.

Contributed Paper

11:05

2aBA8. Ultrasound tomography of materials with high sound speed contrast. Timothe Falardeau and Pierre Belanger (Mech. Eng., Ecole de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C1K3, Canada, timothe.falardeau.1@ens.etsmtl.ca)

Non-destructive evaluation of materials using ultrasounds is frequently used in industry as a way to characterize material properties and to locate defects. Imaging methods based on back propagation of echo signals are limited to reconstruction of low spatial frequency. Ultrasound diffraction tomography is a transmission-based imaging method which gives the possibility of characterizing anisotropic materials. Recent progress in ultrasound tomography using a circular array of transducers enabled velocity mapping of materials with high sound speed contrast relative to the background. In this study, an image of acoustic properties of a titanium rod submerged in water was obtained using bent ray time-of-flight tomography. The experiments were performed on a circular array ultrasonic test bench at 2 MHz frequency on 322 transduction points. A 2D finite element model of wave propagation through water and titanium was developed in order to validate experimental results. Finite element reconstruction matched experimental results within a 10% error for geometry and velocity.

Invited Papers

11:20

2aBA9. Recent advances in resonant ultrasound spectroscopy (RUS) for the measurement of the stiffness tensor of anisotropic and attenuative materials. Quentin Grimal (Biomedical Imaging Lab., Sorbonne Universités - Université Pierre et Marie Curie, 15 rue de l'école de médecine, Paris 75006, France, quentin.grimal@upmc.fr)

The elasticity tensor of a small sample of anisotropic material can be advantageously determined with Resonant Ultrasound Spectroscopy (RUS). In RUS, resonant frequencies of a sample are measured, and computed frequencies of a numerical model of the sample are fitted, yielding the stiffness tensor. RUS was developed in the 1990s, but until recently, it was in practice limited to measure materials with a high quality factor. We have recently adapted the method to measure attenuative materials such as plastics and hard biological tissues (bone and tooth tissues) with a typical quality factor of about 25. Our strategy combines Bayesian methods to retrieve overlapped resonant peaks in the RUS spectra and to solve the inverse problem using a limited number of resonant frequencies. The method allows a quasi-automated processing of RUS spectra where it is not necessary to know a priori the pairing between measured and computed frequencies. In the last years, we have extensively used RUS to document the anisotropic elastic properties of human bone and we explored application in additive manufacturing.

11:40

2aBA10. Optical-resolution photoacoustic imaging with speckle illumination. Emmanuel Bossy (LIPhy, Université Grenoble Alpes/CNRS, LIPhy, 140 rue de la Physique, Saint-Martin d'Hères 38400, France, emmanuel.bossy@univ-grenoble-alpes.fr)

Conventional approaches for optical-resolution photoacoustic microscopy generally involves raster scanning a focused spot over the sample. Here, we show that a full-field illumination approach with multiple speckle illumination can also provide diffraction-limited optical-resolution photoacoustic images. Two different proof-of-concepts are demonstrated with micro-structured test samples. The first approach follows the principle of ghost imaging [1], and is based here on solving a linear inverse problem under sparsity assumptions: the object is reconstructed through a pseudo-inverse computation of a reference matrix made of speckle patterns measured during a calibration step. The second approach is a speckle scanning microscopy technique, which adapts the technique proposed in fluorescence microscopy by Bertolotti *et al.* [2]: in our work, spatially unresolved photoacoustic measurements are performed for various translations of *unknown* speckle patterns. Because speckle patterns naturally appear in many various situations, including propagation through biological tissue or multi-mode fibers, speckle-illumination-based photoacoustic microscopy provides a powerful framework for the development of novel reconstruction approaches, well-suited to compressed sensing approaches. [1] Katz *et al*., "Compressive ghost imaging," *Appl. Phys. Lett*. **95**(13), 2009. [2] Bertolotti *et al*., "Non-invasive imaging through opaque scattering layers," *Nature* **491**(7423), 2012.

Session 2aEA

Engineering Acoustics: Thermophone Transduction

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Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. Platinum films to carbon nano structures: The history of the thermophone transducer. Andrew R. Barnard and Mahsa Asgarisabet (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

The thermophone is a device which creates sound using rapid heat oscillations on its surface. This results in a thin and lightweight loudspeaker with no moving parts. Braun was the first to address this phenomenon in the late 1800s. Arnold and Crandall developed the first theory and correlated experimental results in 1917. Unfortunately, the materials necessary to make an efficient thermophone did not exist in their time. In 2008, Xiao rediscovered the thermophone effect using carbon nanotube thin films, a material much better suited to efficient thermophones. Since then many researchers have been working on developing thermophone technology using carbon nanostructures including nanotubes, nanofibers, and graphene. This talk will cover the history of the thermophone from its early days through today and give a broad overview of the application areas for this technology, which span from underwater transducers to consumer electronics and automotive applications.

9:25

2aEA2. Thermoacoustic sound projector: Beyond the fundamental efficiency of carbon nanotubes. Ali E. Aliev (Alan G. MacDiarmid Nanotech Inst., Univ. of Texas at Dallas, P.O. Box 830688, BE 26, Richardson, TX 75083, Ali.Aliev@utdallas.edu)

Advances in thermophone transduction from the perspective of novel nanostructured materials, device design, and signal processing will be presented. The comparison of studied 2D and 3D networks of nanostructured materials with aerogel structures will be given. The energy conversion efficiency of encapsulated thermoacoustic devices excited by short pulses with varying duty cycle, shape of pulse, and carrier signal frequency will be analyzed for a variety of fabricated devices. I will provide an extensive experimental study of pulse excitation in different thermodynamic regimes for freestanding carbon nanotube sheets with varying thermal inertias (single-wall, multiwall with varying diameters, and number of superimposed sheets) in vacuum and in air. The experimental observations are rationalized within a basic theoretical framework. The acoustical and geometrical parameters providing further increase in efficiency and transduction performance for open and closed resonant systems will be discussed. [Research supported by ONR, Grant No.: N00014-17-1-2521.]

9:45

2aEA3. The fabrication and characterization of nanocarbon foams for their utilization in thermoacoustic device. Mei Zhang, Paul Wolmarans (High-Performance Mater. Inst., Florida State Univ., IME, FAMU-FSU College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, mzhang3@fsu.edu), and Ali E. Aliev (A. G. MacDiarmid NanoTech Inst., Univ. of Texas at Dallas, Richardson, TX)

Thermo-acoustic (TA) sound generation (thermophone) is a non-resonant technique where electrical energy is converted to sound waves through Joule heating of a resistor (TA heat source) without any mechanical vibration, thus allowing for a wideband operation. It is clear that the material and the structure of the resistor play an important role on the performance of the thermophone. Recently, thermophone transducers fabricated from nanoscale materials hold the promise of a new transducer technology. Transducers made from these nanomaterials operate over a broad frequency range and can be designed to be lighter and thinner than competing technologies. Here, we report the TA heat source using nanocarbon foams. Nanocarbon foams are carbon nanotubes (CNTs) based all carbon porous materials. They have a hierarchically porous structure and the pore size and porosity can be tuned easily during the fabrication process. The foams consist of highly porous conductive CNT networks. They are freestanding, flexible, and mechanically robust in various environments. It is demonstrated that the foams can be used as elastically compressible, flexible TA heat source. The detailed fabrication process, the morphology of the foams, their thermal and electrical properties, and their performance as thermophone transducers will be presented.

10:05

2aEA4. Thermophones for sonar applications. Nathanael K. Mayo, Christian Schumacher, Donald Cox, Jeffrey E. Boisvert, John B. Blottman, and Thomas R. Howarth (Naval Undersea Warfare Center: Newport, Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841-1708, nathanael.mayo@navy.mil)

Thermophones are electrically driven thermoacoustic sound projectors which have historically been used as a primary source of sound. Thermophones have a sound spectra that are largely determined by their housing and support and efficiencies that are determined by the device's ability to maintain a dynamic temperature gradient across a gaseous layer surrounding the thin heating element. A number of factors make thermophones an attractive technology for underwater use including the relative ease and low cost of production, the large thermal reservoir provided by the surrounding aquatic environment, and the ability to tune the spectra over a broad range of frequencies. We present calibrated acoustic underwater tests performed on thermophones which demonstrate the potential for a new class of sonar transducers. Small, 6.35 cm diameter, inert gas filled laminate pouch thermophones were fabricated which provide a low frequency resonance. Additionally, a liquid filled thermophone demonstrates a smooth response over a wide frequency band.

10:25–10:40 Break

10:40

2aEA5. Cylindrical heat conduction models for enclosed thermophones. Benjamin Dzikowicz, James Tressler, and Jeffery W. Baldwin (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil)

The efficiency of enclosed thermophones is governed by the time-dependent heat transport between the heating element and the enclosure gas. Efficiency gains have been made by reducing the heat capacitance of the heating elements so that a maximum amount of energy is available for transport. However, efficiency is still limited when there is poor conduction into and out of the enclosure gas. An analytical model for the conduction in an enclosed thermophone with a heating element composed of a regularly spaced linear array of conducting fibers will be presented. The model considers interaction between the individual fibers and treats the enclosure wall as a heat sink. The conduction model will be incorporated into a device model which can calculate steady-state acoustic output. The results are compared against experimental data. In addition, the device model can be used to explore the optimal fiber spacing for enclosed thermophones with regularly spaced linear array heating elements. [Work funded by the Naval Research Laboratory.]

Contributed Papers

11:00

2aEA6. Carbon nanotube coaxial thermophone for automotive exhaust noise cancellation. Suraj M. Prabhu, Andrew R. Barnard (Mech. Eng., Michigan Technolog. Univ., 1400 Townsend Dr., 815 R.L. Smith Bldg., Houghton, MI 49931, smprabhu@mtu.edu), and Mahsa Asgarisabet (Mech. Eng., Michigan Technolog. Univ., Plymouth, MI)

Exhaust noise from automobiles is a major concern and needs to be controlled. Current noise control systems, mainly mufflers, have significant size, weight, and performance limitations at low frequencies. Passive control systems are unable to cancel noise efficiently over the entire frequency spectrum. Active noise control system helps to overcome the above limitation. This paper deals with the design and testing of a coaxial carbon nanotube speaker for active cancellation of automotive exhaust noise. Carbon nanotubes are virtually massless and work on the thermo-acoustic principle. Their ability to sustain heat makes them suitable for use in elevated temperature environments such as automotive exhausts. Analytical calculations are performed using transfer matrix method to evaluate the effect of coaxial thermo-phone and side branch speakers on the sound pressure level in the exhaust tailpipe. Design of the coaxial CNT thermophone and testing of it by mounting on an automotive exhaust tailpipe are discussed. Low frequencies (20 Hz–1000 Hz) are considered to evaluate the performance of the thermophone. For active cancellation of exhaust noise, Filtered-X Least Mean Square (FXLMS) algorithm is developed and implemented using NI LabVIEW.

11:15

2aEA7. An experimentally validated electrical-thermal-acoustic model of non-planar carbon nanotube thermophones. Mahsa Asgarisabet, Andrew R. Barnard, and Suraj M. Prabhu (Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, masgaris@mtu.edu)

Carbon nanotube (CNT) thin-film thermophones are a solid state, transparent, magnet-free, stretchable, and lightweight transducers that work via the thermoacoustic effect. The rapid change in the temperature of the CNT film with low heat capacity produces a temperature wave accompanied by an acoustic wave in a frequency range of 1 Hz to 100 KHz. The existing lumped parameter models for the planar CNT thermophones are not appropriate for the complex geometries of the CNT film. Using COMSOL multiphysics, an electrical-thermal-acoustic model of the non-planar CNT thermophone has been developed. By applying an alternating electrical current to the CNT film, the temperature variation was obtained and used to simulate the pressure distribution. For different input power levels, the temperature distribution on the CNT film was compared to the experimental data from thermal camera. The experimental sound pressure level at different locations in front of the CNT thermophone were recorded and used to validate the non-planar model.

11:30

2aEA8. Application of carbon nanotube thermophones in consumer electronics. Stephania M. Vaglica (Mech. Eng., Michigan Technolog. Univ., 52134 wembly ct, Shelby twp, MI 48315, smvaglic@mtu.edu) and Andrew Barnard (Mech. Eng., Michigan Technolog. Univ., Houghton, MI)

A thermoacoustic transducer is a device that converts electrical energy into heat energy which then generates sound energy. Carbon nanotubes (CNT) are a thermoacoustic transducer that have the ability to transform this energy. Carbon nanotubes are hexagonal lattice tubes of carbon that have diameters one ten thousandth of the thickness of human hair. These graphite allotropes of carbon have a range of different applications and properties, particularly thermoacoustic conductivity. Because of their unique thermoacoustic properties, a lightweight loudspeaker can be designed by creating an aligned film of carbon nanotubes. CNTs form nanoscale tubes that are positioned in a "forest" like configuration. Unlike a typical loudspeaker, there is no magnetic coil that generates sound. Instead, sound is generated by the nanotubes fast cooling and heating property. This cooling and heating effect will expand and contract the air around the speaker which will result in the propagation of a sound wave. This new type of speaker creates a brand new market for speciality consumer electronics.

11:45

2aEA9. Thermal and vibration induced rotor blade deicing using carbon nanotube thermophones. Steven Senczyszyn and Andrew R. Barnard (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, sasenczy@mtu.edu)

The accumulation of ice on helicopter blades imposes limitations on the design and operation of rotorcrafts due to the large power requirement of thermal heating mats. Carbon nanotube (CNT) loudspeakers present a lightweight and efficient solution to this issue, utilizing both thermal and vibratory methods to remove ice and prevent it from forming. To evaluate the deicing capabilities of CNT, an enclosure was created to encapsulate the loudspeaker between two thin layers of aluminum to simulate the rotor blade surface. The speaker was then operated at a range of amplitudes and frequencies in simulated cold weather conditions. Preliminary tests show promising results, with surface temperatures exceeding 120°C in under four minutes when operated in a −20°C environment and surface excitation resultant from the ultrasonic vibration. Although further testing is needed, CNT loudspeakers show great promise in becoming a lightweight and efficient solution to the issue of ice accumulation on the surface of rotor blades.

TUESDAY MORNING, 5 DECEMBER 2017 STUDIOS FOYER, 9:00 A.M. TO 12:00 NOON

Session 2aED

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Murray S. Korman, Cochair
Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402

All posters will be on display from 9:00 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 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

2aED1. Music a scientific art: A call for review of Department of Music, University of Nigeria Nsukka's curriculum and course outline. Stephen G. Onwubiko (Music, Univ. of Nigeria, University of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Every academic discipline ranging from the Social Sciences, Engineering, Physical Sciences, Arts, and Humanities is typically categorized as either a Science or Arts discipline. While Music is artistic and creative, there is harm in designating music as solely an Arts discipline. It is literally true that music is art, yet the viral damage of categorizing music as solely an Art discipline cannot be overemphasized. Just like a "wolf in sheep skin," music is science dressed up like art. Music is an offspring of pure science (physics) expressed as art. Music is made up sound, frequency, pitch, resonance, vibration, echo, and Doppler effect; these topics are pure science with mathematical derived formulas and not just art. For example, the action of a typical upright piano obeys Newton's third law. This paper condemns the outright definition and categorization of music as strictly an art and recommends the re-evaluation of music education at the university. The music curriculum should be rejuvenated and embedded with elective courses from the department of Physics and Astronomy to enable scholars know more about the science of their instruments, while acquiring the mastery of the art of music.

2aED2. Vowel mergers in the American South. Shawn C. Foster, Joseph A. Stanley, and Margaret E. Renwick (Linguist, Univ. of Georgia, 1907 S. Milledge Ave., Apt. D-4, Athens, GA 30605, shawnchristianfoster@gmail.com)

This study investigates the occurrence of a series of vowel mergers in the English of the Southern United States. Investigated mergers include the pin-pen merger, which may be considered characteristic of Southern English, as well as the cot-caught, pull-pool, fill-feel, and cord-card mergers, which are less strongly identified with the region. First and second formant values of over 300,000 vowel measurements from 41 speakers were automatically extracted from.wav files contained in the Digital Archive of Southern Speech (DASS). Vowel distributions were statistically analyzed via Pillai scores, Euclidean distance, and mixed-effects modeling to determine to what degree, if any, the speakers participate in these mergers. Previous studies on vowel merger in the United States have typically classified speakers into discrete "merged" or "unmerged" categories, often relying on speaker or listener judgments to do so. The richness of the acoustic data and high number of tokens available for this study allow speakers to instead be classified along a spectrum, where they are described by "degree of merger" rather than binarily. The results in turn provide new perspective on how Southerners have participated in or resisted dialectal changes that have shaped American English from the late 18th through the mid-20th century.

2aED3. Student experiment for measure of target strength. Zhengliang Cao, Jianfeng Tong, and Wei Shen (Hadal Sci. and Technol. Res. Ctr., College of Marine Sci., Shanghai Ocean Univ., #999 Hucheng Huan Rd., Shanghai 201306, China, zlcao@shou.edu.cn)

The target strength is a measurement of the reflection coefficient of a sonar target. Its value and feature is very important to detection and identification in active sonar, especially for fish by fishery acoustic students. A steel cylindrical shell is used as the target, whose outer diameter 194 mm, inner diameter 184 mm, length 500 mm, and both end closed covers with thickness 4 mm. A broadband transducer is selected to directionally project sound wave and a hydrophone receives the reflection wave from the target. The transducer and receiver are located at 5 m and 4 m from the target with the same depth 5 m in the water. When single frequency pulses are transmitted with signal interval frequency 50 Hz, the target strength is calculated by amplitude ratio of target echo and incidence wave at different frequencies. The result is compared with theoretical value for an infinite elastic cylindrical shell and finite length elastic cylindrical shell. This experiment will give a strategic guidance to understand what target strength is, how to understand it, and why we need it. [Work supported by National Natural Science Foundation of China (Grant No. 41374147).]

2aED4. Evaluating the validity and efficiency of tablet audiometry in school-age children. Jenna Supinski (Health Sci. and Human Performance, Ithaca College, 953 Danby Rd., Ithaca, New York 14850, jsupinski@ithaca.edu), Olivia M. Pereira (Ctr. for Pediatric Auditory and Speech Sci., Nemours/ Alfred I. duPont Hospital for Children, Wilmington, DE), Mackenzie Hammond (Dept of Commun. Sci. & Disord., West Chester Univ., West Chester, PA), Samantha Scharf (Dept of Linguist and Cognit. Sci., Univ. of Delaware, Wilmington, DE), Thierry Morlet, and Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Nemours/ Alfred I. duPont Hospital for Children, Wilmington, DE)

Tablet-based automated audiometry offers a portable alternative to traditional audiometry. The limited research available (e.g., Rourke *et al*., 2016) supports the clinical use of automated audiometry for pediatric hearing screenings, but its accuracy in children and efficiency outside of clinical environments is undetermined. This study's objective is to evaluate the validity and efficiency of automated audiometry in school-age children. This initial phase aimed to establish its validity in a clinical setting. Hearing thresholds for 0.5, 1, 2, 4, 6, and 8 kHz were collected in children ages 6–12 years old using standard audiometry and an iPad app automated audiometry in a sound-proof booth. Tympanometry, acoustic reflex thresholds, and distortion product otoacoustic emissions were administered to examine peripheral hearing function. Preliminary results of 16 children show no significant difference between the two test durations. Automated audiometry thresholds were within 5 dB of the standard audiometry thresholds for each tested frequency, except at 6 kHz where they were within 7dB. There was no test preference among the participants. Our preliminary results support the use of automated audiometry in children. Current testing is evaluating its validity and reliability under less ideal testing environments. [Work supported by NIH COBRE Grant P30GM114736 and the Nemours Foundation.]

2aED5. Assessing temporal compensation of speech due to delayed auditory feedback. Samantha N. Davis (Univ. of Washington, 5521 Summer Blvd., Galena, OH 43021, sndavis94@gmail.com) and Francois-Xavier Brajot (Ohio Univ., Athens, OH)

This study examined the behavioral compensation of speech influenced by delayed auditory feedback. It was hypothesized that delayed auditory feedback would yield similar compensation patterns as other auditory perturbations to speech do, in a hypothesis untested by previous experiments. The results of past studies which experimented with pitch, loudness, and formant frequencies look similar to each other, due to somatosensory feedback's role in adaptation. Compensatory responses, when measured against non-altered productions, seem to approach an asymptote, yielding incomplete compensations. Compensatory responses, when measured against the size of the perturbation, resemble a decreasing exponential function, as relative magnitudes get drastically smaller with increasing perturbation size. The data supported the hypothesis as predicted, meaning the compensation to delayed auditory feedback appears to follow the same principles as those observed in other altered auditory feedback paradigms. These results suggest that this compensation technique is consistent for all altered auditory feedback methods, and that it follows a generalized rule for sensorimotor feedback control in speech production.

2aED6. Implementing electronic speckle pattern interferometry for a variety of activities in a general education acoustics class. Nicholas J. Razo and Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

A familiar acoustics laboratory experiment is finding Chladni patterns using flat plates of various geometries. Another method used in musical acoustics to find mode shapes of vibrating objects is electronic speckle-pattern interferometry (ESPI). We introduced students in a general education acoustics class to ESPI in addition to the traditional Chladni plate laboratory activity. A variety of ways to implement ESPI in the class were developed including: comparing traditional Chladni patterns to ESPI, examining traditional musical instruments with ESPI, and an advanced investigation of a well-known acoustics demonstration. A classic acoustics demonstration is rubbing the rim of a wine glass to create a musical tone. A coffee mug, unlike a wine glass, has an asymmetry due to the mug handle. ESPI can be used to study the effects of filling the mug with water in terms of the mug's natural frequency, amplitude, and mode shapes. Results from ESPI showed that the mode shapes do not change with the addition of water. Filling the coffee mug with water lowered the natural frequencies of the lowest modes in a manner similar to that of a wine glass.

2aED7. Sound speed profile calculations for the Northern Gulf of Mexico. Bradley J. Sciacca (Dept. of Phys., Univ. of New Orleans, 43 Lake Lynn Dr., Harvey, LA 70058, bsciacca@uno.edu) and Juliette W. Ioup (Dept of Phys., Univ. of New Orleans, New Orleans, LA)

Many sound speed calculations for the Gulf of Mexico use an average sound speed over the entire Gulf. In a variety of applications, such as localization or tracking of marine mammals, a more precise sound speed profile is helpful to make more accurate calculations. Using a reasonable approximation to Del Grosso's equation, the sound speed profile may be computed as a function of temperature, pressure, and salinity. The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015 using Environmental Acoustic Recording Systems (EARS), returning to sites previously surveyed by LADC. Oceanographic data were also collected at those sites and can be used for more accurate sound speed profiles at the specific locations and specific times of the acoustic data recordings, increasing the accuracy of subsequent calculations such as real-time tracking and other applications. Sound speed profiles for sites in the northern Gulf of Mexico will be presented. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

2aED8. Noise levels during recreational flight of a Cessna 172 Skyhawk. Drew Jones (Audiol. & Speech Pathol., Univ. of Tennessee, 1144 Wincreek Dr., Collierville, TN 38017, djone120@vols.utk.edu), James D. Lewis, and Mark Hedrick (Audiol. & Speech Pathol., Univ. of Tennessee, Knoxville, TN)

Recreational and private pilots are regularly exposed to potentially hazardous noise levels during flight training sessions and recreational flying. For the first part of this study, light airplane noise was recorded during recurrent 1.5-hour flight training sessions. Measurements were made at the level of the pilot's left and right shoulders (using dosimeters). Data demonstrated that changing inflight operations and power adjustments were associated with specific acoustic patterns. Noise levels (Leq) ranged from 76 dBA to 96 dBA during flight. The second part of this study examined the effect of the noise on subjective and objective measures of the pilot's auditory function. Measurements were made before and immediately after flight training sessions. Pure-tone audiometric thresholds, middle-ear absorbance and click-evoked otoacoustic emissions did not show significant changes from pre- to post-flight. Although inflight noise levels are potentially hazardous, the use of an aviation headset acted to mitigate the risk of hazardous exposure. Future research directions include measurements in diverse aircraft and across longer flight durations.

2aED9. Regional dialect perception by British and American listeners. Emma C. Bonifield (Speech and Hearing Sci., Indiana Univ., 1615 Springhill Rd., Warsaw, IN 46580, embonifi@indiana.edu), Tessa Bent (Speech and Hearing Sci., Indiana Univ., Bloomington, IN), Bronwen Evans, Isla Tyrrell, and Fernanda Aguilera (Univ. College London, London, United Kingdom)

Speakers' regions of origin, across and within countries, can substantially impact their native language pronunciation patterns. Previous work suggests that listeners are sensitive to these dialect differences within their home country (Clopper & Pisoni, 2004) and have at least broad representations of dialects outside of their home country (Bush, 1967). The current experiment expands on these findings by investigating dialect perception within and across multiple countries for listeners from two countries: United States and United Kingdom. Fifty-one talkers from the Speech Accent Archive (Weinberger, 2013) representing 6 U.S. regional dialects, 5 U.K. dialects, and 6 dialects from other English-speaking countries produced the same two sentences. Participants ranked these speech samples on their proximity to "Standard American English" or "Standard Southern British English" in two ladder tasks. American and British listeners' ladder rankings for the American baseline were very highly correlated, with a clear split between North American and non-North American dialects. For the British baseline, American listeners again had a split between the North American and non-North American dialects, while the British listeners showed more gradation across the dialects. The results suggest that listener home dialect and dialect familiarity, possibly through media exposure, shape perceptual representation of regional dialects.

2aED10. Full-spectrum comparison of denoising algorithms for real-time magnetic resonance imaging acoustics. William Klock and Marissa Barlaz (Linguist, Univ. of Illinois at Urbana-Champaign, 707 S Mathews Ave., 4080 Foreign Lang. Bldg. MC-168, Urbana, IL 61801, wklock2@illinois.edu)

Using real-time MRI acoustic data, we employ two methods of signal denoising (DLWP and CS-SNG) to conduct a preliminary comparison between noisy, denoised, and noiseless data. The acoustic data collected in the MRI serve as the noisy, "baseline" group. Data collected from the same speakers in a sound-attenuated environment served as the noiseless, "ground truth." We calculate acoustic power across the frequency spectrum in 32, 64, 128, and 256 bin experiments and perform k-means clustering on the first three principal components to compare the output of the denoising algorithms to the ground-truth and noisy data. Results show a quantitative difference between the denoising methods, through their different affinities for clusters associated with reference group labels. The groupings indicate that the CS-SNG data are better suited for establishing a map between the visual data from the MRI and the acoustic output because of its association with the noiseless data and its distinction from the noisy group. Since both denoising algorithms form independent clusters, there are potential differentiating features that could drive future improvement of these methods.

2aED11. Measurements of the Doppler effect due to a rotating sound source. Elizabeth McQueney, Maryam Landi, and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, mcqueney.e@gmail.com)

The Doppler effect is a common principle in all types of waves when sources and receivers have a relative motion. In acoustics, there are scenarios where sound sources rotate and produce the Doppler effect. These scenarios include sound produced by a speaker with a rotating horn and by the rotational motion of helicopter blades and airplane propellers. In this study, we experimentally measure the frequency shift due to a rotating sound source. A buzzer (2.8 KHz) is set to rotate at a fixed radius (75 cm) with the rotational velocity (60 rpm) controlled by a stepper motor. The radiated sound signals are recorded by a microphone at different locations. Spectrograms of the recorded signals display a shift from the buzzer's original frequency. The shift oscillates with the period of the source rotation. We characterize the dependence of the shift on the position of the microphone. A formula for the shift is derived and is used to compare with the measurements. Our measurements and analyses gain insight into the Doppler effect due to a rotating source.

2aED12. Localizing crow vocalizations in social aggregations. Derek Flett, Virdie Guy, Shima Abadi (Eng. and Mathematics, Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, flettd23@uw.edu), and Douglas Wacker (Biological Sci., Univ. of Washington, Bothell, WA)

The North Creek Wetlands Restoration on the University of Washington Bothell campus is home to a large nocturnal American crow (*Corvus brachyrhynchos*) roost. Each day from Autumn to Spring, crows form pre- and post-roost aggregations, which consist of tens to hundreds of crows. Crows on these aggregations are often highly vocal, but the functions of their vocalizations are not well understood. Identifying any context-dependent patterns in these vocalizations is critical to fully understand communication in this highly social and intelligent species. Previous studies have shown the presence of human observers near large groups of crows may disrupt natural vocal and non-vocal behavior. In this study, the potential confound of these observer effects are eliminated by recording crow vocalizations using a remotely activated, time-synched microphone array. Simulations are undertaken to study the performance of the Time Difference of Arrival (TDOA) method to localize individual callers. A parametric study is used to analyze the effects of number of receivers, signal frequency and duration, and crow location on the performance of TDOA. In addition to the simulation, different types of recorded crow vocalizations are used to design robust playback experiments to fine-tune our localization technique for use in actual crow aggregations in future.

2aED13. Comparisons of measured sound power levels across octave bands gathered from different methods and labs. Samuel H. Underwood and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, samuelunderwood@unomaha.edu)

This poster presents results to date from an interlaboratory study, aimed at quantifying the bias and reproducibility of three different sound power measurement methods used in the heating, ventilation, and air-conditioning industry: free field method, diffuse field method, and sound intensity method. The sound power of a certain loudspeaker sound source has been measured in at least 15 testing facilities using the methods generally applied in those facilities. Two test signals are used in the measurements: a broadband signal whose slope decreases −5 dB per octave band, and the same signal with four discrete frequency tones at 58, 120, 300, and 600 Hz. Comparisons of the measured sound power level data, gathered to date, are presented across one-third octave bands. The project continues to collect data from other test facilities so that rigorous findings on the repeatability, reproducibility, laboratory bias, and measurement method biases may be determined, according to ISO 5725. [Work supported by the Air-conditioning, Heating and Refrigeration Institute.]

2aED14. Modeling nonlinear acoustic landmine detection using a soil plate oscillator apparatus. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com)

In modeling nonlinear acoustic landmine detection, there will be a cylindrical drum-like buried target. This target, which has a clamped thin elastic plate, rigid sidewalls, and bottom plate, is buried in an open concrete "soil tank" containing dry sifted masonry sand. Subwoofers located 40 cm above the surface are driven with a swept tone from 50 to 450 Hz to generate airborne sound that couples into the sand exciting vibration in the buried target. A small geophone is used to measure the "ground" vibration profile across the buried target. Nonlinear tuning curves of vibration particle velocity vs. frequency are used to compare "over—off the target" results. The backbone curves (peak velocity vs. corresponding resonant frequency) are

different in these cases—such that resonances due to soil alone or soil over a compliant buried target can be distinguished. Next, a soil plate oscillator SPO (consisting of two circular flanges sandwiching and clamping a thin circular elastic plate that supports a cylindrical level column of sand above the plate) is driven by airborne sound. Nonlinear tuning curve vibration at points across the sand are compared with the results in the "idealized" landmine detection experiment to develop a lumped element model of the system.

2aED15. Computer simulation of synthetic aperture sonar or radar for classroom demonstration: Part II. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

In our presentation, a Mathematica® simulation of Synthetic Aperture Acoustic Radar will demonstrate how point-like targets placed on a smooth surface can be imaged from a collection of acoustic echoes. The transmitter and receiver are collocated and modeled as a single element that stops along a linear track at collection points and hops to the next location (stop and hop approximation). The transmitter on the hypothetical apparatus will transmit a linear frequency modulated LFM chirp pulse that reflect off the targets and are recorded as echoes by the receiver at discrete locations along the track. A matched filter correlation process will perform a pulse compression of the LFM chirp. A series of still frames from the animated simulation will be displayed. These frames show the echoes arriving at different times from the targets in conjunction with a receiver wave recorder screen. The time correlation backprojection algorithm is motivated when pulse compressed arrivals are displayed (at various track locations) including the echo arrival of a hypothetical point target adjusted in the x,y plane on or near one of the actual point targets. [See Yegulap, "Fast backprojection algorithm for SAR,“ in Proc. 1999 IEEE Radar Conf., Waltham, MA, April 20–22, 1999, 60–65].

2aED16. Investigation of sound level variations in occupied K-12 classrooms. Jared Paine (Civil Eng., Case Western Reserve Univ., 2095 Martin Luther King Jr. Dr., Cleveland, OH 44016, jap198@case.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

Sound level data have been logged in 220 K-12 classrooms, as part of an investigation underway at the University of Nebraska—Lincoln on environmental conditions inside occupied classrooms. In each classroom, equivalent sound levels were logged every 10 s with an integration time of 10 s, during 36 hour periods that spanned two occupied school days, at three different times throughout an academic year. Previous presentations on this dataset have focused on average occupied and unoccupied sound levels for each classroom; this poster investigates the variation of the sound levels in the classrooms throughout the occupied days and how those differ across the grade levels measured (specifically, 3rd, 5th, 8th, and 11th grades). Among the quantifiers analyzed to understand sound level variation are standard deviation, L10-L90, and occurrence rates. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

TUESDAY MORNING, 5 DECEMBER 2017 STUDIO 4, 8:15 A.M. TO 11:15 A.M.

Session 2aMU

Musical Acoustics: Measurement Methods and Instrumentation for Musical Acoustics

Wilfried Kausel, Cochair
Inst. of Music Acoustics, Univ. of Music and Performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Thomas Moore, Cochair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—8:15

Invited Papers

8:20

2aMU1. Detailed experimental modal analysis of a trumpet: An application of laser Doppler vibrometry. Martin Cockrill (ASDC Dept. of Eng., The Univ. of Leicester, UK, ASDC Unit 5, The Control Ctr., MIRA Technol. Park, Nuneaton CV10 0TU, United Kingdom, mc490@le.ac.uk) and Maximilian Chowanietz (ASDC Dept. of Eng., The Univ. of Leicester, UK, Leicester, United Kingdom)

Much work has been conducted on the physics of musical instruments and the trumpet has been included in numerous studies. However, the majority of the work is directed at understanding the air column physics since that is the dominant mechanism of a brass instrument. While this is the largest contribution to understanding the sound, it fails to explain the minute differences between instruments from both the player's point of view and the audience. We have looked to add to the global body of work by conducting a full experimental modal analysis of an entire trumpet. By utilizing a 3D scanning laser Doppler vibrometer, we have been able to measure the dynamic response both in and out-of-plane over the entire trumpet. We have been able to do this at around 500 locations providing detailed geometric resolution and resolving higher order modes. We have then solved the resulting dynamic model to provide natural frequencies and mode shapes. It is hoped that the results of this work can be combined with the virtual modelling of other researchers to refine the state of the art in terms of understanding the properties and mechanisms inherent in trumpets and the quality of the sound they produce.

8:40

2aMU2. Experimental techniques for the visualization of the acoustic fields radiating from brass musical instruments. Amaya López-Carromero, D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, 1608, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh, Scotland/Midlothian EH9 3FD, United Kingdom, s1374028@sms.ed.ac.uk), and Jonathan A. Kemp (Music Ctr., Univ. of St Andrews, St., Andrews, Fife, United Kingdom)

Radiation from brass instrument bells under artificial excitation can be visualized using a number of experimental techniques. In this work, we present two different approaches: Schlieren optical imaging using a high speed camera and acoustic 2D scanning using a linear microphone array. The potentials and limitations of both methods, as well as their methodological requirements, are discussed. Examples of measurements on several instruments are presented. Schlieren optical imaging is applicable for the detection of large amplitude wavefronts formed by nonlinear wave steepening and is this is particularly sensitive to the high frequencies that form within these waves. The 2D scanning linear microphone array on the other hand can, in principle, measure all frequencies and the use of the exponential sine sweep method allows any nonlinear behaviour (in the loudspeaker and microphone or in the air) to be removed to get an accurate measurement of the linear radiation field for any frequency of choice within the measurement bandwidth.

9:00

2aMU3. Flow visualizations using electronic speckle pattern interferometry. Whitney L. Coyle and Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, wcoyle@rollins.edu)

Imaging air flow in and around a musical instrument is a difficult task, but it can be important for understanding the physics associated with sound production as well as verifying the accuracy of computer simulations. Particle image velocimetry (PIV) has been successfully used to image air flow, but it requires expensive equipment, extensive technical expertise, and sophisticated software. Furthermore, PIV is usually restricted to observing only a small area of the flow. As an alternative to PIV, we demonstrate a method based on electronic speckle pattern interferometry that allows large-area real-time imaging of air flow using minimal optical equipment.

9:20

2aMU4. Searching for early reflected waves after strike of Caribbean steelpan using time-resolved electronic speckle pattern interferometry. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu), Thomas Moore, and Daniel Zietlow (Dept. of Phys., Rollins College, Winter Park, FL)

Electronic speckle-pattern interferometry has become a popular tool for studying the operating deflection shapes and normal modes of musical instruments. Normally, ESPI is used with cameras having frame rates below 50 Hz, and the resulting interferograms are the result of a time average over many oscillations. Using ESPI with a high speed camera operating at frame rates of several thousand frames per second allows for time-resolved examinations of transient motion, and this technique has been used to study the motion immediately following the strike on a Caribbean steelpan. Caribbean steelpans are a complex system of coupled oscillators and it has been suggested that the inflections from the concave shape of the steelpan bowl to the convex shape of the note areas serves as a boundary where mechanical waves are partially reflected. High speed ESPI movies of strikes on a low tenor steelpan were acquired in an effort to search for evidence of these reflected waves.

Contributed Papers

9:40

2aMU5. Humidity influences on natural Timpani heads. Wolfgang Nagl and Alexander Mayer (Dept. of Music Acoust. (Wiener Klangstil), mdw – Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, wolf.nagl@gmx.at)

Varying air moisture is an issue timpanists using natural skins have to deal with. Moisture, which in concert halls is often created by the audience or by the musicians on the stage, causes natural skin to loose its tension and therefore flattens the pitch of the instrument. The present paper deals with the effects of moisture on the pitch of timpani with natural skin presenting an experiment for measuring the relationship between air moisture and skin tension resp. pitch variation. An experimental setup has been created where the natural skin head was mounted on a copper shell containing an ultrasonic humidifier. The humidification process was computer controlled in order to establish exact humidity levels. The resulting change of the kettledrum skin parameters was measured using an impulse hammer and force transducers at all mounting points of the tensioning hoop. Stretching parameters have been captured by a special video-system. Measurements have been made in a relative humidity range of up to 91%. The results may also guide musicians and help them to cope with humidity variations during their concerts.

9:55

2aMU6. Sound fields forever: Mapping 3D sound fields using position-aware smartphone technology. Scott H. Hawley, Sebastian Alegre, and Brynn Yonker (Chemistry & Physics, Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu)

Google Project Tango is a suite of built-in sensors and libraries allowing certain mobile devices to track their motion and orientation in three dimensions, without the need for any external hardware. Our new Android app, “Sound Fields Forever,“ combines position information with sound intensity data for multiple frequency bands obtained from a comoving external microphone plugged into the phone’s analog audio jack, and filtered digitally. These data are sent wirelessly via a WebSockets connection to a visualization server running in a web browser. This system is intended for multiple roles: (1) education, providing visual representations of introductory acoustics laboratory phenomena such as interference patterns and room modes; (2) live sound reinforcement, providing front-of-house engineers real-time maps of a loudspeaker system’s effects throughout the venue; (3) architectural acoustics, aiding builders and designers in evaluating acoustical properties of halls. The relatively low cost of our approach (~$400 for the smartphone) compared to more sophisticated 3D acoustical mapping systems could make it an accessible option for such applications. We present visualizations of various sound field measurements and intend to demonstrate how the bandpass filters make it suitable for use even in the presence of nontrivial ambient noise.

10:10–10:30 Break

10:30

2aMU7. Study of the impact of material on clarinet-like instrument: Correlation between impedance measurement and musician tests. Louise Hovasse, Jonathan Cottier, Jerome Selmer (Henri Selmer Paris, PARIS, France), and Vincent Gibiat (ICA Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr)

The acoustical effect of the material of woodwind instruments seems to be the favorite discussion subject between acoustician and instrument maker. Indeed in scientific literature, different hypothesis are demonstrated by musician tests or scientifically experimentation. This study, based on the acoustical input impedance measured on tubes and clarinets, as well as musician tests, tends to highlight material could have part of influence on the acoustic of woodwind instrument. Moreover, correlations between measurements and musician tests are found. Indeed, three clarinets have been tested by musicians, one as reference in African Blackwood, one half synthetic-half African Blackwood with the same density as the reference instrument and one in other species of wood with a lower density than the reference instrument. The correlation between measurement and musicians tests is that the instrument felt more brilliant is the one with the highest quality factors on the impedance peaks and reflection coefficients, whereas the instrument felt duller have the smallest quality factors and reflection coefficients. Since the quality factor and the reflection coefficient are linked to the visco-thermal losses, themselves depending on the surface condition, the acoustic of the clarinet-like instruments appears to be more impacted by the surface condition than the material itself.

10:45

2aMU8. Investigating vocal tract effects during note transitions on the saxophone. Montserrat Pàmies-Vilà (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Inst. 22, Vienna AT1030, Austria, pamies-vila@mdw.ac.at), Gary Scavone (McGill Univ., Montreal, QC, Canada), Alex Hofmann, and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Vienna, Austria)

The resonances of the vocal tract are used by single-reed woodwind players for tuning purposes, timbre modification, and other musical effects. Using pressure or impedance measurements, the influence of the vocal tract on saxophone playing has been previously studied for steady tones. This study considers note transitions in which the players might perform vocal tract modifications (e.g., wide intervals) and combines pressure and reed bending measurements to analyze the effect of the vocal tract as well as its dependence on the articulation technique. The measurement setup consists on recording the acoustic pressure in the players' mouth and in the mouthpiece of the instrument. A strain gauge attached to the reed surface captures both the reed oscillations and the tongue-reed interaction due to articulation. Complementary details about the playing technique are obtained by recording the upper teeth force on the mouthpiece and the open-closed configuration of two selected keys. Preliminary results show that saxophonists are able to drive the vibrations of the reed by modifying the vocal tract configuration, while adapting the blowing pressure and tongue-reed interaction to the articulation technique.

11:00

2aMU9. Preliminary measurements of vibrations and resonances of a *Kundum* xylophone. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The *kundum* xylophone is unique among bar instruments because it has bull horn resonators. The *kundum* is made and played primarily by the Birom ethnic group found in Vom in the Plateau state of Nigeria, which is in the central and northern part of the Nigeria. It complements both the Western and African musical genres but is played mostly in traditional folk music. The unique tone quality of the *kundum* is far more superior and richer than other xylophone types found in traditional Nigerian societies. The handmade *kundum* consists of a row of wooden slabs that are struck with a hammer. The vibrations of the wooden slabs are amplified by closed-tip bull horns placed beneath the slabs. The bull horns are sized to produce a high-pitched note; the *kundum* uses overtones of the open-closed bull horns. This paper examines preliminary measurements of the vibrations of the wooden slabs, as well as natural resonances of closed-pipe bull horns. Comparisons are made with the resonances of ideal cylindrical and conical open-closed tubes to see if the effect of the curved tip of the bull horns warrants further study.

Session 2aPA

Physical Acoustics: Sound Used as an Investigative Tool for Industrial Solutions

Gabriela Petculescu, Chair
Physics, University of Louisiana, Lafayette, 240 Hebrard Blvd., Lafayette, LA 70504

Chair's Introduction—9:00

Invited Papers

9:05

2aPA1. Grain size distribution measurement of Ti-6Al-4V plate using laser-ultrasonics. Feng Dong (National Eng. Res. Ctr. of Flat Rolling Equipment, Univ. of Sci. and Technol. Beijing, 2145 Sheridan Rd., Tech B224, Evanston, IL 60208-3109, dongf515@gmail.com), Quan Yang (National Eng. Res. Ctr. of Flat Rolling Equipment, Univ. of Sci. and Technol. Beijing, Beijing, China), and Sridhar Krishnaswamy (Ctr. for Smart Structures and Mater., Northwestern Univ., Evanston, IL)

As the most widely used titanium alloy, Ti-6Al-4V, has wide range of applications in aerospace industry and medical prostheses. Grain size is a significant parameter that should be controlled during material processing for the reason that it can affect the strength and toughness directly. Laser-ultrasonics is an appropriate method for in-situ measurement of the grain size and distribution during heat treatment of Ti-6Al-4V as it is a nondestructive and non-contact technique. In this study, a laser-ultrasonic system was established combining an 8 ns width pulsed laser for ultrasound generation and a two-wave mixing (TWM) interferometer for detection. Several Ti-6Al-4V samples were heat treated variously to get different grain sizes but with the same phase ratios. Attenuation rates of ultrasonic waves in the different specimens were precisely calculated using exponential fitting for various frequencies. The true values of grain size and distribution were observed via scanning electron microscopy and electron backscatter diffraction (SEM-EBSD). The frequency dependent attenuation of ultrasonic waves was obtained for various grain sizes of Ti-6Al-4V. A statistical correlation between the grain size distribution and the frequency-dependent ultrasonic attenuation was established.

9:25

2aPA2. Ultrasonic characterization of the complex Young's modulus of polymers produced with micro-stereolithography. Clinton B. Morris (Mech. Eng., The Univ. of Texas at Austin, 204 Dean Keeton, Austin, TX 78705, clint_morris@utexas.edu), John M. Cormack, Michael R. Haberman, Mark F. Hamilton, and Carolyn C. Seepersad (Appl. Res. Labs. and The Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Micro-stereolithography is capable of producing polymer parts on the scale of millimeters that contain micron-scale features. Accurate prediction of the dynamic performance of components produced using micro-sterolithography requires knowledge of the as-fabricated material properties, but very little published material property data exist for these materials. This is complicated by the fact that the properties vary as a function of build parameters (i.e., laser exposure time). Designers therefore have limited useful material property information for part design. Frequency-dependent material parameters are often determined by measuring the wave speed and attenuation of an ultrasonic pulse as it propagates through the material. This work employs laser Doppler vibrometry to detect extensional waves in a solid polymer rod of circular cross-section that is excited by a longitudinal ultrasonic contact transducer. Transverse motion associated with extensional waves propagating along the rod axis is measured at multiple surface points. The time series gathered from all locations are used to produce a frequency-wavenumber spectrum. These data, coupled with a forward model that accounts for both viscoelastic and geometric waveguide effects, are then used to invert for the lossy material properties from 0.4 to 1.2 MHz.

9:45

2aPA3. Study of temperature and pressure dependent elastic properties of porous ceramics. Ashoka Karunarathne, Josh R. Gladden, and Gautam Priyadarshan (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38677, atholang@go.olemiss.edu)

Resonant Ultrasound Spectroscopy (RUS) is a widely used experimental approach to investigate the elastic properties of solid materials at different temperature and pressure. In RUS, experimentally obtained vibrational resonance spectrum of a sample is used to calculate the full elastic tensor of the material. We report here a series of RUS measurements of novel porous ceramics material designed by LG which has applications in high temperature fuel cells. The elasticity and the acoustic damping of a porous material are influenced by its porosity and its saturation fluid. The time dependent elasticity obtained from RUS measurements exhibits a two-step process of stiffening with two distinct time constants. The stiffening-softening behavior during the heating-cooling cycles and the reduction of acoustic damping in the low-pressure conditions were observed depending on the synthesis history of the material.

10:05

2aPA4. Acoustic monitoring of aluminum-alloy sensitization. Shankar Kharal (Physics, Univ. of Louisiana, Lafayette, 240 Hebrard Blvd., Lafayette, LA 70504), Nicholas Jones (Naval Surface Warfare Ctr. - Carderock Div., Bethesda, MD), and Gabriela Petculescu (Physics, Univ. of Louisiana, Lafayette, Lafayette, LA, gp@louisiana.edu)

Segregation of atomic species in metastable solid-solution alloys results in a gradual change of the alloy's properties, sometimes rendering the materials ineffective for the particular task they were designed for. An example of such a case is the sensitization of strong and lightweight Mg-rich aluminum alloys used in marine applications. The ASTM standard is an acid-corrosion test (G67), a destructive and lengthy procedure that requires large specimens analyzed off-site. Ultrasound is a well-known tool for nondestructive material characterization and it can offer a solution for on-site testing. To this end, the sensitivity of ultrasonic parameters to the degree of material degradation through sensitization has been identified. Velocity and attenuation for shear and longitudinal waves were measured as a function of sensitization for 5083 and 5456 aluminum alloys with two different methods: Resonant Ultrasound Spectroscopy (RUS) and Pulse Echo (PE). The longitudinal and shear velocities change by 0.5% and 1.5%, respectively, while the attenuation coefficient of longitudinal waves changes by of 20% (all represent saturation values). The JMAK equation for phase kinetics is used to understand the observed evolution of the acoustic parameters vs. sensitization.

10:25–10:40 Break

10:40

2aPA5. Electroacoustic modeling and characterization of an ultrasonic projector. Guy Feuillard, Pascal Tran Huu Hue, Naoufal Saadaoui, Van Thien Nguyen (Insa Ctr. Val de Loire, Blois, France), Marc Lethiecq (Université de Tours, Bat E, 20 Ave., Monge, Parc de Grandmont, Tours 37000, France, marc.lethiecq@univ-tours.fr), and Jean François Saillant (Areva, Chalon/Saone, France)

Ultrasonic projectors are new type ultrasonic transducers that take advantage of the ultrasonic wave emitted on the rear face of the transducer. It consist in an acoustic symmetrical design where both waves generated on the faces of the piezoelectric ceramic are collected in phase with a 45° angle wedge resulting in an increase of the sensitivity. In this case, the absence of backing endows that the conventional design rules of a transducer cannot be directly applied. In this work, the electroacoustic modeling and characterization of such projector is reported. Using the KLM well known model, the electroacoustic response of a projector in based on a piezoelectric ceramic with one matching layer is calculated. Simulations show that the optimal acoustic impedance of the matching layer should be lower than for conventional transducers leading to a bandwidth of approximately 50%. The characterizations of such transducer based on PZ27 (Meggit/Ferroperm) and a lead free material PIC700 (PI Ceramic) have been carried out. Bandwidth and sensitivity are reported. They are found close to the simulation results and demonstrate that these new types of transducers has to be designed according new trends compared to conventional transducers.

11:00

2aPA6. An acoustic approach to assess natural gas quality in real time. Andi Petculescu (Univ. of Louisiana at Lafayette, 240 Hebrard Blvd., Lafayette, LA 70503, andi@louisiana.edu)

The natural gas industry needs fast and robust techniques to monitor the quality of the natural gas, if possible during the flow measurement process. A potential technique relies on extrapolating the dynamic specific heat of the gaseous mixture via measurements at a few frequencies (Petculescu and Lueptow, *Phys. Rev. Lett.* **94**, 238301 (2005); Sensors and Actuators B: Chemical **169**, 121–127 (2012)). At the core of this approach lies a first-principles model for sound absorption and dispersion in polyatomic gases. We will show and discuss the applicability and limitations of this potential technique to predicting the content of nitrogen, oxygen, carbon dioxide, and water in natural gas.

11:20

2aPA7. Investigating the effects of formulation and processing conditions on the mechanical properties of wheat flour noodle dough. Reine-Marie Guillermic, Sébastien O. Kerhervé (Dept. of Phys. and Astronomy, Univ. of MB, Allen Bldg., Winnipeg, MB R3T 2N2, Canada, reine-marie.guillermic@umanitoba.ca), Huiqin Wang (Dept. of Food Sci., Univ. of MB, Winnipeg, MB, Canada), Anatoliy Strybulevych (Dept. of Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada), Dave W. Hatcher (Canadian Grain Commission, Grain Res. Lab., Winnipeg, MB, Canada), Martin G. Scanlon (Dept. of Food Sci., Univ. of MB, Winnipeg, MB, Canada), and John H. Page (Dept. of Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada)

Characterization of the mechanical properties of soft food materials is crucial in the food industry, both for process design and for quality enhancement purposes. The goal of this project is to develop a non-contact on-line quality control technique for use in processing of sheeted products such as Asian noodles. In the case of the Asian noodle industry, composition and work input during the sheeting process are important parameters that influence the mechanical properties of the dough, and as a consequence, final product quality. An accurate and fast determination of noodle properties and how they are influenced by formulation and processing parameters will certainly improve quality control during production. To address this need, we are conducting a full study of noodle dough with ultrasonic techniques. A non-contact ultrasonic technique in transmission is used to assess the mechanical properties of noodle dough sheets and has been evaluated in pilot-plant trials, showing its feasibility for real-time quality control. Additionally, for a better understanding of the material properties, including how both bubbles and the dough matrix are affected by sheeting, contact longitudinal experiments over a wide range of frequencies (0.5 to 12 MHz), along with ultrasonic shear experiments, have been performed.

11:40

2aPA8. Ultrasonic investigation of fatigue, tensile damage, and impact damage in woven glass-fiber fabric reinforced composites with comparison to X-ray tomography. Nico F. Declercq (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France, nico.declercq@me.gatech.edu), Pascal Pomarède (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France), Othmane Ez-Zahraouy (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, Metz, France), Nada Miqoi (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France), Lynda Chehami (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, Metz, France), Fodil Meraghni, and Laurent Peltier (LEM3, UMR CNRS 7239, Art et Métiers Paris Tech, Metz, France)

This work was done on woven glass-fiber fabric reinforced composite samples. Those materials exhibit a complex anisotropic evolution of defects induced by several damage mechanisms. In order to non-destructively evaluate the damage accumulation within this material, a methodology based on the measurement of the complete stiffness tensor is considered. After validation of the detectability of increasing damage state with this method, a new damage indicator is proposed to thoroughly quantify it. Samples were damaged by tensile tests (quasi-static and fatigue) at increasing stress levels along and out-of fibers axis. Afterwards, drop-weight impact is performed to consider several damage situations. Finally, an X-ray tomography is conducted to identify the damage mechanisms as well as the evolution of the void volume fraction. It is shown that this evolution has the same tendency with the ultrasonic damage indicator.

TUESDAY MORNING, 5 DECEMBER 2017 SALON A/B/C, 8:30 A.M. TO 11:45 A.M.

Session 2aSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics: Acoustic Metamaterials I

Christina J. Naify, Cochair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr., Pasadena, CA 91109

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

8:30

2aSA1. On time-dependence of mechanical metamaterials properties. Massimo Ruzzene and Giuseppe Trainiti (Daniel Guggenheim School of Aerosp. Eng., Georgia Inst. of Technol., 270 Ferst Dr., Atlanta, GA 30332, ruzzene@gatech.edu)

The study focuses on wave propagation in time-periodic mechanical matamaterials, such as beams with time-modulated elastic modulus. The time-dependence of the elastic modulus results in a system with a frequency-periodic dispersion diagram. Flat bands originate at the intersections of the dispersion branches, signaling zero group velocity modes. Through an analytic approach, we obtain approximate expressions for the range of wavenumbers associated to flat bands, as well as their dependence on the modulation parameters. It is shown that the flat bands lead to unbounded growth of the displacement field, which is due to the interaction between the structure's dynamics and the energy provided to it by means of the elastic modulus variation. If such phenomenon is properly confined over a limited time window, it provides a way to filter wave propagation over a ultra-wide frequency range by property choosing the modulation parameters. Numerical simulations, obtained by implementing a FDTD scheme, verify the theoretical findings. Finally, an experimental validation of the results is discussed, which employs piezoelectric patches and time-dependent negative capacitance shunted circuits.

8:50

2aSA2. A metamaterial design method applied to topologically protected mechanical metamaterials. Kathryn Matlack (Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 1206 W. Green St., Urbana, IL 61801, kmatlack@illinois.edu), Marc Serra-Garcia (Inst. for Theor. Phys., ETH Zürich, Zurich, Switzerland), Antonio Palermo (Dept. of Civil, Chemical, Environ. and Mater. Eng. - DICAM, Univ. of Bologna, Bologna, Italy), Sebastian D. Huber (Inst. for Theor. Phys., ETH Zürich, Zurich, Switzerland), and Chiara Daraio (California Inst. of Technol., Pasadena, CA)

Mass-spring models provide a straightforward way to describe complex dynamic behavior of mechanical systems, such as topologically protected and backscattering-free surface phonons. Such discrete models also provide a means to translate these properties, based on the unique class of electronic materials of topological insulators, to the mechanical domain. This talk will discuss a systematic design process to take such a mass-spring model and implement its functionality directly into a metamaterial. The design method uses

combinatorial searches plus gradient-based optimizations to determine the configuration of the metamaterial's building blocks that replicates the behavior of the target mass-spring model. A key aspect of the design method is the use of "perturbative metamaterials"—i.e., metamaterials with weakly interacting unit cells. The weak coupling enables the design space to be divided into smaller sub-spaces that can be independently tuned, which results in an exponential speed-up of the design process. It also allows us to isolate the interactions between vibrational modes of its unit cells within the frequency range of interest, simplifying the inverse problem. The design method is illustrated through the example of engineering a 2D mechanical metamaterial that supports topologically protected and defect-immune edge modes, directly from the mass-spring model of a topological insulator.

9:10

2aSA3. Active acoustic metamaterials using sensor-driver architecture. Bogdan Ioan Popa (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

Acoustic metamaterials can, in principle, overcome the fundamental limitations of passive structures and can enable novel applications that are hard or impossible with passive media. However, active acoustic metamaterials remain elusive, and the few implementations going beyond basic material parameter tunability lag significantly behind the true potential of active media. We will present a design method that results in active metamaterials whose acoustic response is electronically programmed to cover a very large range of non-linear acoustic responses. Specifically, metamaterials obtained with this approach can be configured to have almost any second and higher harmonic response, making them very effective and versatile non-linear acoustic media. We demonstrate experimentally this design methodology in an application in which an acoustically thin metamaterial funnel responds to incident sound waves by producing collimated second harmonics. Past demonstrations of this type of active metamaterials will also be briefly reviewed.

Contributed Papers

9:30

2aSA4. Design and measurement of an acoustic bi-anisotropic metasurface for scattering-free manipulation of the refracted wavefront. Junfei Li, Chen Shen (ECE, Duke Univ., 101 Sci. Dr., Rm. 3417, FCIEMAS Bldg. Duke Univ, Durham, NC 27708, junfei.li@duke.edu), Ana Díaz-Rubio, Sergei Tretyakov (Dept. of Electronics and NanoEng., Aalto Univ., Aalto, Finland), and Steven Cummer (ECE, Duke Univ., Durham, NC)

Relying on the generalized Snell's law which defines a phase gradient to the surface, extraordinary control of the reflected and transmitted wavefronts can be achieved by metasurfaces with subwavelength thickness. However, a fundamental limitation for such metasurfaces is their power transmission efficiency, especially at large deflection angles. Building on the theoretical requirements for the boundary conditions at an ideal metasurface, we designed and fabricated the necessary bi-anisotropic cells for a wave-front transformation acoustic metasurface that overcomes the fundamental limits of conventional designs, allowing us to steer the energy flow in an arbitrary manner without parasitic scattering. The design with a discretized structure is verified both numerically and experimentally and an energy conversion efficiency of 93% is observed for the transmitted wave into the desired direction of 60 degrees, higher than the corresponding theoretical upper limit in generalized Snell's law based designs (89%). Our experimental demonstration confirms the theoretical results and opens a new way of designing practical and highly efficient metasurfaces for different functionalities, allowing nearly ideal control over the energy flow through metasurfaces.

9:45

2aSA5. Passive acoustic metasurfaces for high-efficiency wavefront transformations. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

Metasurfaces have offered new degrees of freedom to control the propagation properties of acoustic waves. Gradient metasurfaces, first proposed in optics, and recently extended to acoustics, add an extra transverse momentum to the incident plane wave so that reflected and refracted waves can be rerouted without having to rely on conventional Snell's Law. However, recent studies have shown that gradient metasurfaces cannot transfer all incident energy into the desired direction without causing unwanted parasitic diffraction and limiting the overall transformation efficiency. More sophisticated metasurface designs have considered impedance matching to address this issue, but this inherently requires the use of active and lossy materials for extreme wave transformations. In this talk, we present our recent efforts in designing passive acoustic metasurfaces with unitary efficiency in beam steering, even when considering extreme steering angles. We consider nonlocal surface impedance profiles and the use of bianisotropic inclusions, which are both effective routes to realize efficient metasurfaces without relying on gain elements. Our work opens new design approaches for metasurfaces, and it can improve the operational efficiency of metasurfaces.

10:00

2aSA6. Anomalous scattering effects from bi-anisotropic inclusions and metamaterials. Li Quan and Andrea Alu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 1616 Guadalupe St., UTA 7.215, Austin, TX 78712, liquan@utexas.edu)

The acoustic scattering properties of small particles are typically dominated by weak monopolar or dipolar effects, associated with their nature and geometry. As large collections of them form materials, these features determine the overall response to sound of the object we have around. Recent work on acoustic bi-anisotropic acoustic inclusions has shown that, by breaking geometrical and time-reversal symmetries, one can realize constitutive relations that couple pressure fields to dipole radiation and velocity field to monopole radiation in non-trivial ways. In this talk, we use these properties to induce anomalous scattering properties in small inclusions, including unidirectional scattering cancellation, non-reciprocal scattering responses, and directive or sharp scattering resonances. We also consider the use of active acoustic particles to further extend the available degrees of freedom in scattering engineering, and explore unusual parity-time symmetric responses in acoustics. By combining several of these complex inclusions to form acoustic metamaterials, we also investigate the unusual material responses offered by this design approach, which goes beyond what available in natural materials, offering new degrees of freedom for acoustic engineering.

10:15–10:30 Break

10:30

2aSA7. Asymmetric transmission via lossy gradient-index metasurfaces: Role of diffraction. Chen Shen (Elec. and Comput. Eng., ECE, Duke Univ., Durham, NC 27705, chen.shen4@duke.edu), Yong Li (Inst. of Acoust., School of Phys. Sci. and Eng., Tongji Univ., Shanghai, China), Yangbo Xie, Junfei Li (ECE, Duke Univ., DURHAM, NC), Yun Jing (Mech. Eng., North Carolina State Univ., Raleigh, NC), and Steven Cummer (ECE, Duke Univ., Durham, NC)

The development of acoustic metasurfaces has enabled numerous wave control abilities. The effect of losses in acoustic metasurfaces for sound transmission manipulation, however, is largely unexplored. Here, we show that robust asymmetric transmission can be achieved by harnessing

judiciously tailored losses. Theoretical investigations show that the asymmetric behavior stems from loss-induced suppression of high order diffractions. Multiple reflections occur inside the individual slits for the negative incident direction, which lead to different orders of diffraction and asymmetric responses for opposite directions. Numerical simulations based on effective medium are performed and are in good agreement with theoretical analysis. Real structures based on unit cells with designed internal viscous loss are fabricated and measured in a 2D waveguide. The peak energy contrast is about 10 times in a certain range of incident angles and frequencies. This study may open up new possibilities in lossy acoustic metamaterials and metasurfaces. The theory can also be readily extended to electromagnetic waves.

10:45

2aSA8. Optimal attenuation caused by boundary impedance in acoustic waveguides. Matthew Kelsten and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu)

Non-Hermitian systems can exhibit "exceptional points" (EPs) at which modes coalesce. The connection between EPs and acoustic damping goes back to the observation of Cremer (1953) that optimal attenuation in a duct occurs when the two lowest modes have equal complex-valued eigenvalues, although the physical basis for this effect remains unclear. In an attempt to understand Cremer's observation, we consider the model case of a two-dimensional waveguide with different impedance conditions on the two boundaries. This allows us to determine the complete set of all possible pairs of passive impedance conditions that give rise to EPs, and from these to select impedances appropriate to a particular frequency bands. The non-separable, and generally non-symmetric, mode shapes are described. The theoretical findings are linked to realistic passive impedance values based on various models for boundary impedance, such as perforated and porous panels. These comparisons are discussed to illustrate the feasibility of optimized wall impedances in absorbing sound passing through ducts. [Work supported by NSF.]

11:00

2aSA9. Thermoviscous effect on sound propagation in acoustic metastructures. Likun Zhang, Xudong Fan (Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu), Xue Jiang (Nanjing Univ., Nanjing, China), and Yong Li (Tongji Univ., Shanghai, China)

Thermoviscous dissipation plays an important role in sound propagation through acoustic metastructures for a variety of applications. The thermoviscous effects rely on the resonances in the structures of subwavelength features. Our analyses and numerical simulations reveal that thermoviscous dissipation can significantly reduce the transmission of sound waves through acoustic metastructures of hybrid resonances in certain scenarios even when the thickness of the viscous boundary layer is much smaller than the width of the slit in the structure [Jiang *et al.*, *J. Acous. Soc. Am.* (2017)]. To further explore how the resonances affect the significance of the losses, we analyze and simulate sound propagation in acoustic metamaterials of various configurations, including a slit connected to one or multiple resonators at the side of the slit or at the end of the slit. We take into account both wall friction and thermoviscous diffusivity in both the slits and the resonators. The results are compared with that without the resonators to gain insight into the role of the resonances. Optimal designs for both minimal dissipation and maximal absorption are addressed.

11:15

2aSA10. Acoustics of locally bilinear periodic metamaterials. Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, alexey.titovich@navy.mil)

The scope of traditional metamaterials is expanded by analyzing bilinear discontinuities in an otherwise continuous medium. This amplitude independent nonlinearity is sought for its practicality and utility. Wave propagation has to be solved for in the time domain due to the difference in energy transfer from compression to rarefaction across an interface. Various sizes, orientations and shapes of the discontinuities are analyzed on a unit cell level. Periodic chains with the same type of material discontinuity have recently been shown to exhibit a rich dispersion behavior, such as negatively sloped dispersion branches [Titovich (2017), *J. Acoust. Soc. Am.* **141**(5), 3698]. These waves are now investigated in a continuum, where the discontinuity is modelled as displacement-dependent complex impedance relating the interface tractions. The onset of chaotic behavior as well as its effect on absorption is discussed. Also, the ability of such periodic discontinuities to produce nonlinear non-reciprocity is investigated.

11:30

2aSA11. Closed-cell hyperelastic elements with mechanical instabilities and structural negative stiffness. Stephanie G. Konarski and Michael R. Haberman (Appl. Res. Labs. & Dept. of Mech. Eng., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

Metamaterials are subwavelength structures that generate enhanced or exotic properties that are unattainable with conventional materials. One such class of metamaterials utilizes inclusions with mechanical instabilities to generate a non-monotonic pressure versus strain response and thus regions of structural negative stiffness. By embedding these sub-wavelength structures in a lossy matrix material, it is possible to tune and enhance material properties of the mixture such as acoustic and elastic nonlinearity and energy dissipation. Previous theoretical work on this topic by the authors has focused on a dilute concentration of these types of hyperelastic inclusions within a fluid or nearly incompressible elastic matrix material to determine the overall response via both quasi-static and dynamic nonlinear homogenization methods. The present work focuses on numerical analysis using the finite element method for closed-cell elements that display non-monotonic pressure-strain behavior. The objective is to design sub-wavelength, hyperelastic inclusions that are easy to embed in a fluid or polymeric background material in an effort to generate elevated parameters of nonlinearity or increased capacity to dissipate vibro-acoustic energy. Emphasis is placed on structures that can be fabricated using additive manufacturing techniques. [This work was supported by the Office of Naval Research.]

Session 2aSC

Speech Communication: Articulatory and Acoustic Characteristics of Nasalization

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Chair's Introduction—9:00

Invited Papers

9:05

2aSC1. Speaker and production characteristics of posterior nasal fricatives. David J. Zajac (Univ. of North Carolina at Chapel Hill, CB# 7450, UNC Craniofacial Ctr., Chapel Hill, NC 27599, david_zajac@dentistry.unc.edu)

Posterior nasal fricatives (PNF) are unusual articulations produced by children both with and without velopharyngeal (VP) anomalies to replace oral fricatives. PNFs are produced by occluding the oral cavity and forcing airflow through a partially closed VP port to create turbulent noise. Often, extra noises occur due to displacement of mucous and/or tissue flutter. The headset of the nasometer was used to record the separate oral and nasal acoustic signals from 18 children who produced PNFs. They ranged in age from 4 to 15 years, 11 were males, and 11 had some form of cleft palate. Examination of the oral acoustic waveforms revealed that all children produced a stop gesture during intended fricative targets—typically a lingual-alveolar or palatal articulation. Sixteen of the children produced a clear flutter (raspberry-like) noise as evidenced by quasi-periodic components in the nasal spectra. All children used PNFs to replace /s/, 13 (72%) to replace /z/, 12 (67%) to replace /tʃ/, and 6 (33%) to replace /ʃ/ and /dʒ/. Most of the children had known histories of conductive hearing loss. Most of the children with cleft palate had adequate VP closure during stop consonant production. Possible causes of PNFs are discussed.

9:25

2aSC2. The relation of auditory perceptual ratings of nasality to nasal port area in connected speech. Brad H. Story and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Modeling can be used to understand the relation of articulatory movement to the acoustics and perception of speech. Published studies of velopharyngeal orifice size and perceived nasality in clinical populations have not shown a clear relationship between the measures. Two reports based on computational modeling, however, have found a high correlation between ratings of nasality and nasal port area (Bunton & Story, 2012; Bunton, 2013). These studies were based on sustained vowel simulations with nasal port areas ranging from 0 to 0.5 cm^2. Expert listeners were able to detect nasality in vowel samples with a nasal port area greater than 0.01 cm^2 and nasality rating plateaued with areas greater than 0.16 cm^2. The present study extends this work by reporting on nasality ratings based on short connected speech samples simulated with varying nasal port areas. The connected speech samples included either obstruent consonants or approximant consonants to examine whether the relationship between perception and nasal port area varies based on the pressure demands of the speech sounds produced. This work allows for a more complete explanation of the relation between auditory perceptual ratings of nasality and nasal port area. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

9:45

2aSC3. Using oropharyngeal articulation to compensate for nasalization: Acoustic and perceptual evidence. Panying Rong (Speech-Language-Hearing: Sci. & Disord., Univ. of Kansas, 1000 Sunnyside Ave., Lawrence, KS 66045, prong@ku.edu), David P. Kuehn (Speech and Hearing Sci., Univ. of Illinois, Champaign, IL), and Ryan Shosted (Linguist, Univ. of Illinois, Urbana, IL)

A variety of oropharyngeal articulatory differences have been identified between oral and nasal vowel pairs in different languages, which either enhance or reduce the acoustic effects of nasalization. The aim of this study is to simulate oropharyngeal articulatory adjustment strategies to maximally compensate for the acoustic effects of nasalization. Two articulatory models were used for the simulation—one speaker-independent (SI) model and one speaker-dependent (SD) model. The results suggested that (1) the oropharyngeal articulatory strategies generated by both models effectively reduced the acoustic effects of nasalization on the low-frequency spectrum of the vowels; and (2) the oropharyngeal articulatory strategies generated by the SD model provided a better compensatory effect in the spectral region of F2 and higher formants compared to the strategies generated by the SI model. Moreover, a comparison of nasality

ratings of the synthetic nasal vowels with and without oropharyngeal articulatory adjustments showed a significant reduction of nasality following oropharyngeal articulatory adjustments. These findings provide acoustic and perceptual evidence in support of motor equivalence. With further research on other speech sounds and the temporal pattern of nasalization warranted, the findings might have potential clinical implications for reducing hypernasality in persons with resonance disorders.

10:05

2aSC4. Articulatory insights on the evolution of nasal vowels in Slavic. Ryan Shosted (Linguist, Univ. of Illinois at Urbana-Champaign, 4080 FLB MC-168, 707 S Mathews Ave., Urbana, IL 61801, rshosted@illinois.edu)

The complex articulatory-acoustic mapping of the vocal tract suggests that the acoustic effects of nasalization are tied to other articulatory mechanisms besides velopharyngeal opening; this has been borne out in many recent studies. But do these findings settle on any consensus regarding the articulatory mechanisms (besides velopharyngeal opening) that differentiate nasal and oral vowels and cause shifts in vowel inventories across time? Do articulatory results mirror or contradict the relatively larger corpus of findings regarding the acoustics of nasalization? Are these results idiosyncratic reflections of language differences or do they represent universal properties of phonological development? To approach these questions, results of a series of magnetic resonance and electromagnetic articulography studies of French, Portuguese, and Hindi are compared to historical sound changes in Slavic, a language family where most nasal vowels are now lost. Nasal vowels have been implicated in a number of diachronic Slavic vowel shifts, particularly the lowering of nasal /i/ and nasal /e/ and the raising of nasal /a/ in the development of Late Middle Slavic. Height-related phenomena are well-attested in the articulatory studies and seem predictive of Slavic changes; however, frontness changes, also attested in the articulatory studies, do not find clear Slavic counterparts.

10:25–10:45 Break

Contributed Papers

10:45

2aSC5. Articulatory correlates of phonemic and coarticulatory nasalization. Marissa Barlaz, Zhi-Pei Liang, and Brad Sutton (Linguist, Univ. of Illinois at Urbana-Champaign, 707 S. Mathews Ave., MC 168, Urbana, IL 61820, goldrch2@illinois.edu)

Phonological theory distinguishes nasal and oral vowel counterparts by velopharyngeal port opening, neglecting other phonetic differences between phonemic and coarticulatory nasalization. Recent articulatory work provides evidence of oropharyngeal distinctions, in addition to velic lowering. This study (12 Brazilian Portuguese speakers) uses real-time MRI to investigate oropharyngeal differences between oral, phonemically nasal, and phonetically nasalized vowels /a, i, u/. Tissue boundaries in midsagittal vocal tract images were automatically detected to reveal each vowel repetition's aperture function. Principal Components Analysis determined vocal tract regions responsible for the greatest variance in the data. Time-dynamic analyses of vocal tract area in these regions used smoothing spline ANOVA. Results show the tongue body and/or hyperpharynx as the most important articulators. For /a/, nasal vowels demonstrate wider hyperpharyngeal and narrower tongue body regions compared to oral vowels. For /u/, oral vowels show wider hyperpharyngeal and narrower tongue body regions. For /i/, nasal vowels demonstrate wider tongue body and narrower hyperpharyngeal regions. Nasalized vowels manifest apertures intermediate between oral and nasal vowels for /a/ and /u/, and similar to oral vowels for /i/. Results are largely in line with expected acoustic effects of nasalization, and demonstrate that phonetic differences exist between phonemic and coarticulatory nasalization.

11:00

2aSC6. Spontaneous nasalization after glottal consonants in Thai: An rt-MRI investigation of rhinoglottophilia. Sarah E. Johnson (Linguist, Univ. of Illinois at Urbana-Champaign, 508 S First St. Apt. 403, Champaign, IL 61820, sjohnson2052@gmail.com), Brad Sutton (BioEng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Zhi-Pei Liang (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The affinity between glottal and nasal articulations ("rhinoglottophilia") facilitates spontaneous nasalization. In Thai, low vowels nasalize after /h/ and to a lesser degree after glottal stop. Nasalization after /h/ may occur because breathiness and nasalization share high energy at low frequencies and a raised first harmonic. Glottal consonants generally may cause nasalization because aerodynamically they do not require a closed velopharyngeal port. We investigated whether Thai vowels after /h/ and glottal stop exhibit similar degrees of velopharyngeal opening (VPO) and compared these results with acoustic measures of nasalization / breathiness. We calculated nasalization / breathiness by the energy ratio of low and high harmonics; we measured VPO by processing oblique real-time magnetic resonance images of the velopharyngeal port. Four Thai speakers (two females, two males) produced relatively large VPO and high acoustic nasalization / breathiness after /h/. Female speakers nasalized vowels after glottal stop, though they produced overall less VPO and lower acoustic nasalization / breathiness when compared to vowels after /h/. In Thai, vowels after /h/ exhibit more physiological nasalization than vowels after glottal stop; furthermore, low VPO is associated with low acoustic nasalization / breathiness. We conclude VPO is primarily responsible for impressions of nasalization in this context.

11:15

2aSC7. Nasal rustle: An evidence-based description. Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu), Liran Oren (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Ann W. Kummer (Speech-Lang. Pathol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), J. P. Willging (Velopharyngeal Insufficiency Clinic, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and Suzanne E. Boyce (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

"Nasal rustle" (also known as "nasal turbulence") is a loud nasal distortion that can be heard during speech production in certain children with velopharyngeal insufficiency (VPI). It occurs when there is a leak of airflow into the nasal cavity through a small velopharyngeal opening. While the perception of audible nasal emission—including nasal rustle—is a standard means for the diagnosis of VPI, there is no consensus on the sound generation mechanism. Current hypotheses include aerodynamic turbulence, velar flutter, and bubbling of mucus secretion. This study investigates the correlation between the acoustic signal of nasal rustle and physical movement at the superior velopharyngeal port. Several pediatric VPI patients were recorded via high-speed video nasopharyngoscopy and simultaneous nasometry during production of speech sounds susceptible to nasal rustle. Instances of perceived nasal rustle in the acoustic signal were identified and compared with the high-speed video. A high correlation was found between the bubbling frequency of mucus secretion above the velopharyngeal port and frequencies in the nasal acoustic signal, suggesting that secretion bubbling is a mechanism for generating nasal rustle. From this analysis, an integrated description of nasal rustle invoking physiological, physical, and acoustic principles is proposed.

Session 2aSP

Signal Processing in Acoustics and Underwater Acoustics: Detection, Classification, Localization, and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) I

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Chair's Introduction—7:55

Invited Papers

8:00

2aSP1. Detection in an uncertain underwater waveguide. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu) and Ali Abdi (Elec. and Comput. Eng., New Jersey Inst. of Technol., Newark, NJ)

Detection of signals in oceanic waveguides is complex because of uncertainties in the underwater channel and potentially suboptimal location of receiving phones. Accurate detection and estimation are a direct result of optimization with respect to frequency, environmental parameters, and receiver location. We here study the impact of these parameters on detection and propose methods for improvement, including detection processor calibration for optimal performance. We complement our work by investigating detection performance with real data collected at different sensors in a pool. In addition to conventional hydrophones, we also use vector sensors. The rationale behind using a vector sensor is that particle velocity components may show different noise and channel characteristics such as a different number of modes and different dispersive behavior, which are tightly related to detection. We study several detectors and evaluate their performance. [Work supported by ONR and NSF.]

8:20

2aSP2. Achieving the transparent ocean. Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pstein@scisol.com)

Our common impression is that radar systems monitor the vast majority of the airspace above the earth's surface. Safe travel is nearly taken for granted, and for the purposes of national security there is a constant close eye on the comings and goings through our airspace. The same cannot be said for under the ocean's surface, from way out at sea, to near our coast, and within our harbors and waterways. The need for the long-coveted "Transparent Ocean" is evident, and in this paper we explore the visionary aspects of achieving this goal. This includes reviewing concepts and a framework around which state-of-the-art often stove-piped techniques in underwater acoustics, signal processing, and acoustical oceanography, along with non-acoustic modalities including radar and EOIR, are combined into an integrated system.

8:40

2aSP3. Effect of beamwidth on detection of near-bottom targets with multibeam echosounders. Christian de Moustier (10dBx LLC, PO Box 81777, San Diego, CA 92138, cpm@ieee.org)

In multibeam echosounding, beamwidth and sidelobe levels constrain detection of echoes from near-bottom targets that are often masked by echoes from the bottom received in the mainlobe of individual beams or through their sidelobes. Acoustic backscatter data collected with multibeam echosounders from various manufacturers indicate that the customary half power point (−3 dB) metric for the angular width of individual beams underestimates the masking effects of bottom echoes. A better fit of the detected bottom echo trace is obtained with beamwidths estimated 10 dB below the maximum response of each beam. The corresponding beamwidths are roughly 66% larger at −10 dB than at −3 dB for a canonical sinc squared beam pattern. Therefore, using the manufacturers' nominal −3 dB beamwidth specifications yields underestimates by the same percentage of the area ensonified by the pulse within a beam, and corresponding overestimates of the detected acoustic backscatter level used to infer target strength or bottom backscatter coefficients.

Contributed Paper

9:00

2aSP4. The research of a frequency-domain adaptive matched filter.
Juan Hui (Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn)

The most classical detector of active underwater acoustic is the matched filter (MF), which is the optimal processor under ideal conditions. Aiming at the problem of active sonar detection, we propose a frequency-domain adaptive matched filter (FDAMF) with the use of a frequency-domain adaptive line enhancer (ALE). The FDAMF is an improved MF. In the simulations in this paper, the signal to noise ratio (SNR) gain of the FDAMF is about 18.6 dB higher than that of the classical MF when the input SNR is −10 dB. In order to improve the performance of the FDAMF with a low input SNR, we propose a pre-processing method, which is called frequency-domain time reversal convolution and interference suppression (TRC-IS). Compared with the classical MF, the FDAMF combined with the TRC-IS method obtains higher SNR gain, a lower detection threshold, and a better receiver operating characteristic (ROC) in the simulations in this paper. The simulation results show that the FDAMF has higher processing gain and better detection performance than the classical MF under ideal conditions. The experimental results indicate that the FDAMF does improve the performance of the MF, and can adapt to actual interference in a way.

Invited Papers

9:15

2aSP5. The hybrid Cramer-Rao bound of direction finding by a uniform circular array of isotropic sensors that suffer stochastic dislocations. Zakayo N. Morris, Kainam Thomas Wong (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., BC 606, Hung Hom KLN, Hong Kong, zakayo.morris@connect.polyu.hk), Dominic M. Kitavi (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., Kowloon, HungHom, Hong Kong), and Tsair-Chuan Lin (Dept. of Statistics, National Taipei Univ., New Taipei City, Taiwan)

Consider azimuth-elevation direction finding by a uniform circular array of isotropic sensors. In the real world, the sensors may dislocate from their nominal positions. These dislocations could be modeled as random variables having an a priori known distribution. This paper investigates how the dislocations would affect azimuth-elevation direction finding by deriving the corresponding hybrid Cramer-Rao bounds. Maximum a posteriori estimators are derived and Monte Carlo simulations are conducted to validate the derived hybrid Cramer-Rao bounds.

9:35

2aSP6. Angle estimation using a moving PV sensor: Performance assessment. Edmund Sullivan (EJS_Consultants, 46 Lawton Brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

It is well known that bearing estimation performance, using a moving line array of pressure sensors, can be enhanced by using a Kalman filter to jointly estimate the bearing and source frequency. This is because the bearing dependence of the Doppler can be exploited when the source frequency is known. However, it is also known, based on an observability analysis, that it cannot be done using a single moving pressure sensor. Here, it is shown that if a single moving pressure-vector sensor is used, both the bearing and roll angle can be estimated. If only the vector sensor is used, estimation of these angles can still be done, but the performance is poor. The addition of the pressure sensor allows the motion to be exploited, thus significantly enhancing the performance. This phenomenon is shown theoretically by using a Bayesian Cramer-Rao lower bound calculation. An example using simulated data is shown where the improvement is clearly demonstrated.

9:55–10:10 Break

10:10

2aSP7. Intelligent active sonar via tuning of transmit waveform and detection threshold. Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr. MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu) and Steven Schoenecker (Naval Undersea Warfare Ctr., Newport, RI)

We consider active sonar systems that translate high-level tasks to a set of parameter adaptations and arbitrate among competing demands, thereby incorporating cognitive processing in the system. We propose adopting the goal-driven autonomy (GDA) architecture presented in Klenk *et al*. [*Computational Intelligence*, May 2013] to realize intelligent active processing. Using the GDA architecture, the sonar system uses its observations to inform how system parameters are tuned to achieve a set of surveillance goals. In addition, the system is able to reason about its actions, identifying discrepancies between predicted and observed performance and modifying both parameters and goals accordingly. In this work, we focus on a tracking task and consider the transmit waveform and detection threshold as tunable parameters. We describe an active sonar system that tunes these two parameters based on observations of the physical environment and target characteristics, as well as current goals. The intelligent system uses its observations of the environment to generate estimates of target Doppler and of clutter density to tune waveform and detection threshold, respectively. Through simulation, we demonstrate the performance improvement achieved by the cognitive system when tracking a maneuvering target that is traveling through spatially non-uniform clutter.

10:30

2aSP8. Littoral Acoustic Demonstration Center passive acoustic localization and tracking of marine mammals in the Northern Gulf of Mexico. Juliette W. Ioup (Dept. of Phys., Univ of New Orleans, New Orleans, LA 70148, jioup@uno.edu)

The Littoral Acoustic Demonstration Center (LADC) consortium, founded in 2000, has collected underwater acoustic data in the northern Gulf of Mexico since 2001 using Environmental Acoustic Recording Systems (EARS) buoys. These hydrophone systems were developed by the U.S. Navy, and originally LADC could record underwater signals only up to 11.7 kHz. Since then the EARS have been upgraded several times and currently record up to 192 kHz, thus allowing recordings of not only sperm whale clicks and codas, but also signals of beaked whales and dolphins. One goal of LADC is to develop an acoustic method for identification of individual animals, with the goal of an "acoustic catalog" similar to the existing photographic catalog of individual whale flukes. Localization and tracking is an important component for acoustic identification. Some localization and tracking techniques and procedures developed and used by LADC will be presented and discussed.

10:50

2aSP9. Incorporating image sources in the time-domain beamforming model for the localization of sound sources in reverberant rooms. Alice V. Lam, Murray Hodgson (Dept. of Mech. Eng., Univ. of Br. Columbia, 6250 Appl. Sci. Ln., Vancouver, BC V6T 1Z4, Canada, avl@alumni.ubc.ca), and Vincent Valeau (Institut PPRIME, Université de Poitiers-ENSMA-CNRS, Poitiers Cedex 9, France)

In this study, we use a small hemispherical microphone array to locate impulsive or stationary sound sources in reverberant rooms. A modified version of the conventional time-domain beamforming model was used, which incorporates physical information from the early specular reflections of the sound source, as well as the geometric and absorption characteristics of the room. In other words, the propagation model takes into account the potential image sources up to a given image order. We show that the technique can be seen as a time-reversal process, the measured pressure field being simultaneously re-emitted from both the actual array position and its virtual image positions. We then present results applying this modified model to simulated data (generated using an image-source propagation model), as well as measurements in real rooms, using both impulsive and continuous broad-band noise. In all cases, the modified model shows an improvement in localization over conventional beamforming. The issue of appropriately adjusting the image order and mixing time when performing the processing for optimal results is also addressed.

11:10

2aSP10. Detection localization and tracking of aircraft using Sparse distributed passive acoustic system. Alexander Sedunov, Hady Salloum, Alexander Sutin, Nikolay Sedunov (Maritime Security Ctr., Stevens Inst. of Technol., One Castle Point on Hudson, Hoboken, NJ 07030, asedunov@stevens.edu), and David Masters (Sci. and Technol. Directorate, DHS, Washington, DC)

Stevens Institute of Technology has conducted extensive long-term testing of acoustic systems designed to track low-flying small aircraft in remote location and recorded over 2 years of data. The system consisted of 4 nodes placed in difficult remote terrain with separation ranging 1–4 km, each node comprising a pyramid-shaped volumetric cluster of 5 microphones an embedded computer, and a pan-tilt-zoom camera steered to detected targets in real time. and communication device. Each nodes' computer performed direction of arrival finding communicated to a central computer collected that data and processed it to generate tracks and classify targets. The duration and the scale of the deployment allowed to identify and solve many problems, including the effects of propagation delays between station and on cooperative localization and tracking, the seasonal changes in environmental noise, persistent and transient noise sources, and the diversity of targets of opportunity and their signatures. The propagation delay effects led to development of separate trackers for review of target trajectories and for immediate action such as steering the camera. An overview of the algorithms is presented along with the long-term observations. [This work was funded by DHS's S&T Directorate.]

Contributed Papers

11:30

2aSP11. Inference regarding the state of mobile underwater object from a small aperture vertical array. Abner C. Barros and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu)

A narrowband source ensonifies an area of interest with an oncoming submerged target in a refractive undersea environment. A vertical linear hydrophone array is employed to infer the depth, range, and speed of the target by resolving a direct path and one interacting surface path. Tracking the target by means of a continuous wave transmission is challenging due to the difficulty of inferring the frequencies and angles of the two returned closely spaced wave vectors. The complex propagation of sound in a refractive medium presents additional challenges for inversion of the wave vectors to range and depth. A Gibbs sampler is employed to construct the posterior joint density of all parameters taking full advantage of the analytic tractability of the conditional densities of the received amplitudes and phases and of the ambient acoustic noise power. The conditional densities of the ordered wave vectors however are constructed numerically by 2-dimensional inverse quantile sampling. The inferred joint posterior density of the target state is obtained by constructing an inverse transformation of the acoustic propagation model that accounts for the depth dependent sound speed. Simulation results demonstrate the approach at received signal to noise ratios below −3dB.

11:45

2aSP12. Compressive spectral estimation methods using a noise reference. Alexander S. Douglass and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu)

Compressive beamforming methods have shown promise for use in high resolution direction finding and source localization. However, these methods often fail in scenarios with low SNR, particularly if only a few snapshots are available. The Spectral Estimation Method (SEM) (Blacodon and Elias, 2004, J. Aircraft **41** (**6**), 1360–1369) provides a means of source localization and source power estimation by minimizing the mean squares error between cross-spectral-density matrices of a measured signal and modeled signals at each location of interest. SEM With Additive Noise (SEMWAN) (Blacodon, 2011, Applied Acoustics, **72**, 11–21) does the same but also utilizes the cross-spectral-density matrix of one or more noise references. In this presentation, the performance of a compressive version of SEMWAN is assessed for use in low SNR scenarios when the solution is sparse and a noise reference is available. Compressive SEMWAN is analyzed first using far-field, free-space simulations with varying SNR and multiple sources. Compressive SEMWAN is then applied to laboratory measurements taken with real noise sources at varying SNR and including some multipath propagation. The performance in each scenario is compared directly to conventional beamforming, a standard compressive beamforming method, and traditional SEMWAN. [Sponsored by NAVSEA through the NEEC.]

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques I

David P. Knobles, Cochair
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Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Invited Papers

8:20

2aUW1. An overview of the Seabed Characterization Experiment 2017. Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu) and David P. Knobles (KSA, LLC, Austin, TX)

A multi-institutional, multi-disciplinary ocean acoustics experiment, Seabed Characterization Experiment 2017 (SCE2017), was conducted 95 km south of Martha's Vineyard, MA, USA, in March and April of 2017. Three research vessels and more than a dozen principle investigators conducted several types of experiments to improve our understanding of the acoustics of fine grained sediments, forward modeling of sound propagation in shallow water over fine grained sediments, and inverse and statistical inference processes used to characterize the seabed in such environments. The measurements included impulsive and tonal source tows received on vertical and horizontal line arrays, direct measurement of sediment bulk acoustic waves and interface waves, and supporting oceanographic observations. The 2017 experiment was preceded by survey cruises in 2016 and 2015 in which more than 200 sediment cores were obtained and detailed sub-bottom profiling was conducted. Here, an overview of the 2017 experiment is presented. [Work supported by ONR.]

8:40

2aUW2. Sound speed and attenuation in muddy sediments from low-frequency acoustic measurements. Lin Wan, Mohsen Badiey (Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu), D. P. Knobles (KSA, LLC, Austin, TX), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas a Austin, Austin, TX), and Justin Eickmeier (Univ. of Delaware, Newark, DE)

While there have been numerous theoretical and experimental studies on the properties of marine granular sands, there are significantly fewer studies on sediments classified as muds. The validity of geoacoustic models for muddy sediments has not been successfully tested due to the lack of inverted low-frequency sound speed and attenuation from acoustic measurements. The ONR-sponsored Seabed Characterization Experiment (SBCE), conducted in a mud patch on the New England continental shelf in the spring of 2017, provides an opportunity to make substantial improvements in understanding the physical mechanisms controlling sound propagation in muddy sediments. Acoustic signals (e.g., 31g explosive and combustive source signals) detonated at various ranges, depths and azimuths were measured in SBCE. This paper utilizes these measured signals to extract the acoustic normal mode characteristics including modal dispersive curve with Airy phase structure, modal amplitude, modal attenuation coefficient, and mode depth function. These normal mode characteristics are used in geo-acoustic inversion algorithms to estimate low-frequency sound speed and attenuation in muddy sediments as a function of frequency. The performance of different inversion methods using different normal mode characteristics is discussed. [Work supported by ONR Ocean Acoustics.]

9:00

2aUW3. Vector sensor observations of the 2 + mode pattern on the New England Mud Patch at 57 Hz. 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. Dept. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

We present recordings made by an Intensity Vector Autonomous Recorder (IVAR) deployed on the seafloor during the Sediment Characterization Experiment (SCE17) conducted on the New England Mud Patch [40°28' N, 70°35' W] in March of 2017. IVAR continuously and coherently records four channels of acoustic data, three from a tri-axial accelerometer embedded in a neutrally buoyant sphere and one from an omnidirectional hydrophone positioned 10 cm above the centroid of the sphere 1.2 m above the seafloor. Here we focus on IVAR recordings of a 57 Hz continuous wave tone generated by a low-frequency acoustic source (J-15) that was towed at 30 m depth by the R/V Endeavor along transects roughly parallel and perpendicular to bathymetric contours. Over the entire tow track deep signal fades occur in 1000 m intervals, suspected to be caused interference between two modes trapped by the sediment basement. A signature in the vector data identifies an additional interference structure superimposed on this pattern (a 2 + mode pattern at 200 m intervals) which fades when the source moves to the northeast of IVAR, where sediment surveys indicate thinning of the overlaying mud layer. Comparison with a mode based model provides an estimate of the sediment properties.

9:20

2aUW4. The Intensity Vector Autonomous Recorder (IVAR) as deployed in the Sediment Characterization Experiment. Peter H. Dahl and David R. Dall'Osto (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

The IVAR system (Intensity Vector Autonomous Recorder) is a bottom deployed system developed for first-use in the Sediment Characterization Experiment (SCE17), conducted on the New England Mud Patch [40°28' N, 70°35' W] in spring 2017. IVAR continuously and coherently records four channels of acoustic data, three from a tri-axial accelerometer embedded in a neutrally buoyant sphere (diameter 10 cm) and one from an omnidirectional hydrophone positioned 10 cm above the centroid of the sphere positioned 1.2 m above the seafloor. Despite operations being significantly impacted by two Gale-force storms, IVAR obtained 72 h of data over 2 deployments that included low frequency signals from a towed source (J-15) deployed from the R/V Endeavor and broad band SUS charges deployed from R/V Neil Armstrong. Interesting features of the intensity (Umov) vector field emerge in both bandwidth regimes, but here we focus on measurements of the 100+ SUS charges. The measured Umov vector for each SUS charge is first compared with the known source bearing, and is followed by an analysis of precursor arrivals and dispersive water-borne arrivals in the context of the received vector field. The source locations provide complementary information on range and azimuth dependence of the sediment properties.

9:40

2aUW5. Statistical inference for sediment parameters using ship noise and a horizontal array. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), David P. Knobles (KSA LLC, Austin, TX), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper considers the use of broadband noise from ship sources of opportunity in statistical inference for geoacoustic parameters of a layered seabed, with applications to data collected with a bottom-moored horizontal array in the 2017 Seabed Characterization Experiment conducted on the New England Shelf. Statistical inference via a maximum entropy—Bayesian approach that determines the most appropriate model parameterization (number of seabed layers) and provides uncertainty estimates of all model parameters (sediment geoacoustic profiles) is applied. The geoacoustic information content of noise due to large commercial ships in a nearby shipping lane and of noise due to a research vessel traversing the array is quantified and compared. Parameter estimates from the inversions are compared with direct measurements from sediment cores and other geophysical data collected in the experiment area.

10:00–10:15 Break

10:15

2aUW6. Bayesian-Maximum Entropy method applied to seabed geoacoustic models using broadband and narrowband acoustic measurements. D. P. Knobles (KSA, LLC, P.O. Box 27200, Austin, TX 78755, davidknobles@gmail.com), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas a Austin, Austin, TX), William S. Hodgkiss (MPL, Scripps Inst. of Oceanogr., La Jolla, CA), Lin Wan, and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE)

Conditional probability distributions $P(\boldsymbol{H}|\boldsymbol{D},\boldsymbol{M})$ are computed using a Bayesian-Maximum Entropy method applied to acoustic measurements collected during the Seabed Characterization Experiment in the spring of 2017. Marginal distributions are computed for both seabed geoacoustic and source parameter values. A prior range-dependent seabed model $\boldsymbol{M}$ is derived from CHIRP survey measurements made in 2015. The prior bounds of parameter values forming the hypothesis vector $\mathbf{H}$ are obtained from an extensive set of sediment core measurements made in the region. The acoustic data, $\boldsymbol{D}$, were produced by small explosive and combustive sources and towed tonal sources. Two acoustic models are employed to sample the N-dimensional $\boldsymbol{H}$ space: Range-dependent parabolic equation RAM and a normal mode method that includes shear wave effects. The dimensionality of $\boldsymbol{M}$ is optimized via a Gaussian Mixture Model (GMM) and compared to the Bayesian Information Criterion (BIC) and the Akaike Information Criterion (AIC). [Work supported by Office of Naval Research.]

10:35

2aUW7. The measurement of muddy seabed properties using ambient noise coherence. Dieter A. Bevans (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), David R. Barclay (Oceanogr., Dalhousie Univ., Dept. of Oceanogr., Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca), and Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The autonomous passive acoustic lander Deep Sound was deployed five times during the Seabed Characterization Experiment and collected ambient noise data on four hydrophones, arranged in a inverted "T" shape, with three spaced in the horizontal and two in the vertical. The lander was deployed with the bottom-most phones approximately 30 cm above the seafloor, recording over an acoustic bandwidth of 5 Hz–30 kHz. Pressure time series, vertical and horizontal noise coherence (directionality), and the local temperature and conductivity were recorded continuously for periods of 9 hours. Wind-driven surface ambient noise coherence was used to estimate bulk acoustic seabed properties. An analytical Pekeris-waveguide noise model was fitted to the data in order to determine the bulk sound speed, density, and frequency dependent attenuation in the bottom fluid half-space. [Research supported by ONR.]

10:55

2aUW8. Hybrid geoacoustic inversion method and its application to different sediments. Zhenglin Li and Renhe Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 Beisihuan West Rd., Beijing 100190, China, lzhl@mail.ioa.ac.cn)

Bottom acoustic parameters have large effects on sound propagation both in shallow water and in deep water with an incomplete sound channel. A hybrid geoacoustic inversion method is proposed to invert for sound speed, density and attenuation coefficient based on the facts that the bottom acoustic parameters have different sensitivities to sound field at different ranges. The hybrid inversion method is applied to inversion for bottom parameter with different sediments in the Yellow Sea, East China Sea, and South China Sea and verified by using the core sampling measurements. The relationships between the sediments types and the bottom acoustic parameters are given. The differences of the sediment acoustic parameters between sandy silty(or silty sand) and clay(or mud) sediment at low frequency are discussed in the end. [Work supported by the National Natural Science Foundation of China under Grant No. 10434012 and Grant No. 41561144006.]

Contributed Papers

11:15

2aUW9. Path losses in sandy sediments with upward refracting sound speed profiles. Alexander G. Soloway (Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698, alexander.soloway1@gmail.com) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, Seattle, WA)

In this work, receiver-to-receiver path losses calculated from measurements of underwater explosions in shallow water (water depth ~15 m) off the coast of Virginia Beach, VA will be presented. Within the approximate frequency range of 40 to 70 Hz, these data have path losses that suggest energy increasing with range by up to 10 dB. Above this frequency range, these data show energy decreasing with range as expected. An analytical model [Frisk, George V., "Determination of sediment sound speed profiles using caustic range information," *Bottom-Interacting Ocean Acoustics*. Springer, US, 1980, 153–157] and numerical propagation models will be used to show that these path losses can be attributed to an upward refracting sound speed gradient in the sediment. This upward refraction of sound waves propagating in the seabed results in the formation of caustics that define shadow zone boundaries. Below the cutoff frequency of the waveguide, receivers in the shadow zone will exhibit lower sound levels than receivers positioned farther from the source and outside of the shadow zone. While frequencies above cutoff will also undergo refraction in the seabed, the observability of this effect will be masked by trapped modes propagating in the water.

11:30

2aUW10. A full-field scattering coefficient for rough interfaces and a modified sonar equation for long-range reverberation. Anatoliy Ivakin (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A full-field perturbation approach [A.N. Ivakin (2016), *J. Acoust. Soc. Am.*, **140**(1), 657–665] allows modifying two terms of sonar equation for reverberation in complex environments and waveguides with rough interfaces. The first one describes a two-way transmission loss for the full-field intensity in the vicinity of the interface. The second term is a full-field scattering coefficient related to but different from the conventional scattering cross-section per unit area. In particular, the coefficient has a non-zero limit value at small grazing angles of incident and scattered waves defined by the contrast of acoustic parameters at the interface and the power spectrum of roughness. The modified sonar equation allows extremely fast estimating the long-range reverberation based on only one (central) frequency calculation of the transmission loss averaged over a certain span of ranges defined by the used frequency bandwidth, and only one value of the scattering coefficient (at zero grazing angles). No multiple calculations of the reverberation pressure field resulted from Monte-Carlo simulations of random rough interfaces are required. Numerical analysis of ocean reverberation is performed considering different types of rough interfaces: sea-surface, water-sediment interface, buried sediment interfaces, and bottom basement. Model-data comparisons are presented based on results of a recent shallow water reverberation experiment TREX2013. [Work supported by ONR.]

11:45

2aUW11. Azimuthal dependence of sound propagation due to seabed variability in shallow water. Mohsen Badiey, Lin Wan, Justin Eickmeier (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Newark, DE 19716, badiey@udel.edu), David P. Knobles (KSA LLC, Austin, TX), and Preston S. Wilson (Mech. Eng., Univ. of Texas, Austin, TX)

In shallow water regions, the environment has complicated temporal and spatial variability including changes of bathymetry, sediment layer structure, bottom property, and physical oceanographic spatial and temporal changes due to processes like internal waves. All these effects can influence sound propagation in the waveguide. The azimuth angle dependence of sound propagation has been studied using the broadband acoustic signals measured at the Atlantic Generation Station site on the New Jersey continental Shelf, where two distinctive geologic/geoacoustic regions exist [*J. Acoust. Soc. Am.* **96**(6), 1994]. The current paper revisits this idea by analyzing the modal dispersion of broadband acoustic signals deployed along circular tracks at the site of the Seabed Characterization Experiment 2017, where the seabed shows strong azimuthal dependent sub-mud layer ridges overlaid by a relatively uniform mud layer with variable thickness along different directions. The results of this paper can be utilized to assess the azimuthal dependence of the sound propagation in the mud patch region. [Work supported by ONR Ocean Acoustics.]

Session 2pAA

Architectural Acoustics and, Psychological and Physiological Acoustics: Perceived Diffuseness II

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Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Greene Building, 110 8th St., Troy, NY 12180

Chair's Introduction—1:15

Invited Papers

1:20

2pAA1. Novel scattering coefficient for characterizing diffusely reflecting surfaces. Jean-Dominique Polack (Institut d'Alembert, Sorbonne Universités/UPMC, 4 pl. Jussieu, Paris 75242 Cedex 05, France, jean-dominique.polack@upmc.fr) and Hugo Dujourdy (Institut Langevin, PSL Res. Univ., Paris, France)

In a recent publication (Acta Acustica **103**(2017) 480–491), the authors developed an innovative model for describing diffuse sound fields. Developing the diffusion equation one step further, this model makes use of the full stress-energy tensor to provide supplementary relations between sound intensity and sound energy. In the case of non-Sabine spaces (narrow or flat rooms), the supplementary relations naturally introduce new boundary conditions that can be interpreted as scattering coefficients on the walls, in analogy to the traditional absorption coefficients associated with the boundary conditions of the energy equation. The paper explains how the novel coefficient is obtained, compares its significance with the widespread diffuse field wall diffusion coefficient, and proposes an experimental technique for measuring it. Preliminary results will be presented, the expected range of values for the scattering coefficient explained as well as its link to wall impedance in the case of a locally reacting boundary.

1:40

2pAA2. Finding an appropriate headphone-based stereo track playback methods by referencing acoustic characters of professional recording studios. Myoung woo Nam (Dept. of Trans-disciplinary Studies, Seoul National Univ., D406, Iuidong 864-1, Yeongtonggu, Suwon, Gyeonggi 443-270, South Korea, mnam@snu.ac.kr)

Most Red Book CD or streaming 2-track music mixed and mastered in a professional recording studio. However, headphone listener experience different to its original studio condition. The recent headphone has decent frequency response, but the sounds do not match especially in soundstage (width and depth of music mix in a horizontal plane) perspective. Although Recording studios are designed to minimize unwanted reflections and resonances, every studio has its own room characteristics of natural room frequency response. In other words, "the acoustical room characteristic" added on "finalized mix" is the real representation of the mastered 2-track music. In this paper, we tried to find the ultimate headphone-based listening methods by referencing the professional recording studios.

2:00

2pAA3. Perceptual evaluation of a virtual acoustic room model. Gabriel Vigliensoni (Music Res. Dept., McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A1E3, Canada, gabriel@music.mcgill.ca), David Romblom (Dysonics, San Francisco, CA), Marc-Pierre Verge (Appl. Acoust. Systems, Montreal, QC, Canada), and Catherine Guastavino (School of Information Studies, McGill Univ., Montreal, QC, Canada)

Two experiments were conducted to perceptually evaluate a virtual acoustic room model implemented using a virtual microphone technique for the direct path and reflections, the image-source model for the reflection geometry and a Feedback Delay Network reverberator. Experiment 1 aimed to determine the best equalization profile for different room sizes. Anechoic recordings of 7 solo instruments were processed to generate 5 room sizes with profiles corresponding to 5 different absorption settings. Fourteen sound engineers and musicians were asked to rate the resulting stimuli in terms of how realistic each setting was for the instrument in that space. The highest ranked absorption setting for each room size was selected for Experiment 2 to investigate whether listeners could identify different rooms sizes across different instruments. Using a free sorting task, 15 musicians were presented with 7 different solo instruments in 5 different room sizes and asked to group them by type of space in which the instruments were performed. The analysis of the dissimilarity matrices indicates that participants were able to organize sounds by room sizes, but the two largest rooms tended to be clustered together. The results show that the virtual acoustic model is capable of rendering realistic room effects.

Contributed Papers

2:20

2pAA4. Room acoustics: Idealized field and real field considerations. Ernesto Accolti and Fernando di Sciascio (Instituto de Automática, National Univ. of San Juan - National Sci. and Tech. Res. Council, Av Libertador San Martín oeste, San Juan, San Juan 5400, Argentina, eaccolti@inaut.unsj.edu.ar)

How is an acoustically diffuse field defined? To what extent are valid the equations of diffuse field theory? These are the questions addressed in this presentation. The answers are explained through more general theories, in turn explained with figures instead of formulae. The starting point is the idealization of diffuse sound field, from where the basic calculation tools used in architectural acoustics are derived. Then, we go through the physical-mathematical models of wave theory and ray theory assuming diffuse field simplifications and analyze the scope of diffuse field models. Wave models and ray models are presented in a simple format with visual support and reference to the underlying mathematical models. The criteria used to define a diffuse field in frequency domain as well as in temporal domain are analyzed. Finally, we present a review of several state of the art tools used to address the real cases when diffuse field cannot be assumed.

2:35

2pAA5. Development of highly diffuse surfaces from architectural concept to listening experience. Gregory A. Miller, Carl Giegold, John T. Strong, and Laura Brill (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

Many different surface shapes can scatter sound effectively, but not every surface can scatter sound beautifully. When employed over large areas within performance spaces, reflections off sound diffusive surfaces must maintain much of the timbre of the original signal in order to preserve a sound quality pleasing to both performers and audience members. At the same time, these surfaces must be developed in ways that express the architect's visual aesthetic of the room. This paper present means of evaluating and optimizing the sound diffusive capabilities of surfaces within performance spaces, from architectural concept through iterative computer modeling processes and ultimately to physical mockups that can be evaluated both objectively and subjectively.

2:50–3:05 Break

3:05

2pAA6. Energetic wave equation for diffuse sound fields. Jean-Dominique Polack (Institut d'Alembert, Sorbonne Universités/UPMC, 4 pl. Jussieu, Paris 75242 Cedex 05, France, jean-dominique.polack@upmc.fr), Hugo Dujourdy (Institut Langevin, PSL Res. Univ., Paris, France), Baptiste Pialot (Institut d'Alembert, Sorbonne Universités/UPMC, Paris, France), and Thomas Toulemonde (Impedance, Paris, France)

The diffusion equation for modeling diffuse sound fields was proposed some fifty years ago on heuristic principles as an extension to Sabine's diffuse field model, and still receives much attention. In a recent publication (Acta Acustica **103** (2017) 480–491), the authors developed the model one step further, using the full stress-energy tensor to provide the missing relations between sound intensity and sound energy. This introduces some extra terms that, in case of non-Sabine spaces (narrow or flat rooms), can be defined with the help of the boundary conditions in terms of absorption and scattering coefficients on the walls. Integrating the divergence of the stress-energy tensor across the shortest dimensions of the space leads to a propagation equation of the telegrapher type, which can be solved using finite difference time domain simulation. Schemes for one-dimensional (corridors) and two-dimensional (open-space) spaces are proposed, and numerical results compared to measurements in real spaces. The comparison makes it possible to evaluate the absorption and scattering coefficients by an adjustment procedure. The paper discusses the range of values taken by these coefficients and compares them to more traditional building-acoustical coefficients. It also presents under which further assumptions the diffusion equation is recovered.

3:20

2pAA7. Diffusion: When phase and energy can become more important than directivity to the perception of "space". Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This paper will present the newest research on the properties of diffusion that are becoming more important than just directivity to the perception of "space." This includes phase response and energy levels remaining after the act of diffusion.

3:35

2pAA8. A spherical harmonics basis for quantifying the isotropy of sound fields in reverberant enclosures. Mélanie Nolan, Jonas Brunskog, and Cheol Ho Jeong (Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, melnola@elektro.dtu.dk)

This study examines an experimental method for evaluating isotropy in reverberant sound fields, based on an analysis in the spherical harmonics domain. The methodology relies on estimating the wavenumber (or angular) spectrum of the sound field in the room, to characterize the magnitude of the waves arriving from definite directions at the observation point. Subsequently, the obtained wavenumber spectrum is expanded into a series of spherical harmonics, and the multipole moments from the spherical expansion are used to characterize the isotropy of the sound field. This spherical harmonic basis is best suited for characterizing isotropy, as it provides an unequivocal characterization of the symmetry of the wave field. The work examines how theoretical considerations compare with experimental results obtained in various rooms with diverse diffuse field conditions. The experimental results are based on automated measurements using a scanning robot. In addition, the corresponding spatial distribution of the active sound intensity field is determined, making it possible to benchmark the proposed methodology with direct observations of the energy flows in the sound field.

3:50

2pAA9. Factors differentiating the 22.2- and 2-channel reproduced sound fields through an acoustic modeling of three listening rooms. Madhu Ashok (Univ. of Rochester, 500 Comput. Studies Bldg., P.O. Box 270166, Rochester, NY, mashok@ur.rochester.edu) and Sungyoung Kim (RIT, Rochester, NY)

We have simulated two loudspeaker configurations (22.2- and 2-channel reproduction) using the CATT-Acoustic software and analyzed the influence of room acoustics on the perception of multichannel-reproduced music. With the rapid growth of virtual reality (VR) and mixed reality (AR), there is a need for an immersive acoustic system that can maintain compatibility of auditory impressions in various acoustic conditions. The research question of the authors' project is whether an increased number of reproduction channels would reduce the room-induced perceptual difference. To answer this question, we have analyzed physical characteristics from calculated impulse responses (IRs) of three distinct room models (varying dimensions and reflecting surfaces). Among many characteristics, the early decay time (EDT) and clarity (C_{80}) values covary with the loudspeaker configurations. The IRs calculated from a 22.2-channel reproduction system had different EDT and C_{80} values for all three rooms. The change was more evident in a reverberant room (a normal listening room) than a relatively dry room (such as studio control room), possibly due to boosts of the ratio of direct sound to late reflections. No salient factor associated with room differences was observed. The subsequent subjective evaluations, performed by eleven listeners, support that they weighted the perceptual difference associated with the reproduction format more over all other perceptual dimensions.

Session 2pAB

Animal Bioacoustics, Signal Processing in Acoustics, Underwater Acoustics, and Acoustical Oceanography: In Memory of George Ioup: Acoustics in the Gulf of Mexico II

Natalia Sidorovskaia, Cochair
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David K. Mellinger, Cochair
Coop. Inst.for Marine Resources Studies, Oregon State University, 2030 SE Marine Science Dr., Newport, OR 97365

Contributed Papers

1:00

2pAB1. Decadal assessment of the sperm whale population trends in the northern Gulf of Mexico using acoustics. Kun Li, Natalia Sidorovskaia (Physics, Univ. of Louisiana at Lafayette, Broussard Hall, Rm. 103, 240 Hebrard Blvd., Lafayette, LA 70503, kxl1737@louisiana.edu), Christopher Tiemann (R2Sonic LLC, Austin, TX), and Azmy S. Ackleh (Mathematics, Univ. of Louisiana at Lafayette, Lafayette, LA)

Passive acoustic monitoring data have been used to assess population trends of sperm whales in the northern Gulf of Mexico. In this paper, the variability of sperm whale abundance in the Mississippi Canyon area derived from acoustic data collected between 2001 and 2017 is discussed in relation to seasons, habitat type, ambient noise levels, and environmental disturbances. The results have shown that sperm whales were present in the region throughout the entire monitoring period with lower activity in winter months. A considerable habitat shift was observed after the 2010 oil spill with sperm whale activity higher at the sites further away from the spill site. The results clearly indicate the importance of long-term spatially distributed acoustic monitoring in characterizing changes in Gulf of Mexico marine mammal population and their habitat.

1:15

2pAB2. Comparing performance of bottom-moored and unmanned surface vehicle towed passive acoustic monitoring platforms for sperm whale detection. Sakib Mahmud, Natalia Sidorovskaia, Kun Li (Physics, Univ. of Louisiana at Lafayette, 611, W Taft St., Lafayette, LA 70503, sxm3227@louisiana.edu), Chris Pierpoint (BioSci. Group, Seiche, Ltd., Devon, United Kingdom), Christopher Tiemann (LLC, R2Sonic, Austin, TX), and David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Newport, OR)

Passive acoustic monitoring (PAM) is a more efficient method of monitoring the distribution and abundance of deep-diving cetaceans than conventional visual surveys. Many species produce identifiable acoustic signals during echolocation and communication which makes it possible to identify and classify based on their acoustical cues. Three different PAM platforms recorded data in overlapping time periods in the vicinity of the 2010 oil spill site: unmanned surface vehicle towed array, bottom-moored buoys, and Seaglider-mounted hydrophone. The detection rate of unmanned surface vehicle towed array and bottom-moored buoys were compared for their efficiency in detecting marine mammals. Detection events were obtained using independent detectors for each platform and then compared by feeding data through a common detector. The results of this study aid in the development of cost-efficient PAM methodology for mitigation and environmental impact assessment purposes. [This research was made possible by a grant from The Gulf of Mexico Research Initiative.]

1:30

2pAB3. Species-level classification and clustering of beaked whale echolocation recordings. Jack G. LeBien and Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, UNO, New Orleans, LA 70148, jlebien@uno.edu)

The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) consortium has collected passive acoustic monitoring data in the northern Gulf of Mexico since 2001. Recordings were made in 2007 near the Deepwater Horizon oil spill, which have provided a baseline for an extensive study of regional marine mammal populations in response to the disaster. Beaked whales are of particular interest as they remain one of the least understood groups of marine mammals, and relatively few abundance estimates exist. Efficient classification and clustering algorithms are demanded for mining the long-term passive acoustic data. Three algorithms using k-means, self-organizing maps, and spectral clustering are tested with various features of detected echolocation transients. Several methods are observed to effectively isolate recorded echolocation clicks of regional beaked whale species. The waveform fractal dimension is introduced as a feature for marine biosonar classification and shown to improve accuracy. A feedforward neural network classifier is also evaluated and shows performance with high accuracy under various noise conditions. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

1:45

2pAB4. Calculating sperm whale lengths in the Northern Gulf of Mexico. George Drouant and Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, george.drouant@selu.edu)

Sperm whales (*Physeter macrocephalus*) produce short duration acoustical clicks while diving to search for food, each click composed of several pulses. The time interval between consecutive pulses, the interpulse interval (IPI), can be used to estimate the length of the whale. IPIs have been estimated from whales of unknown orientation by processing several hundred clicks and averaging the results. In this work, discrete wavelet transforms and autocorrelation-based methods are used to obtain IPI estimates with improved consistency. The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015. Results presented here are produced using data recorded by the LADC Environmental Acoustic Recording Systems (EARS) at the site closest to the BP oil spill. Calculated lengths can be compared to lengths from

previous measurements as well as to future measurements in order to determine changes in average whale size, useful in determining population health. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

2:00

2pAB5. Dolphin target detection processes investigation using finite element model. Chong Wei, Matthias Hoffmann-Kuhnt (Acoust. Res. Lab., Tropical Marine Sci. Inst., National Univ. of Singapore, 12A Kent Ridge Rd., Singapore 119222, Singapore, weichong3310@foxmail.com), Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kailua, HI), Darlene R. Ketten (Dept. of Otology and Laryngology, Harvard Med. School, Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), and Abel Ho (Acoust. Res. Lab., Tropical Marine Sci. Inst., National Univ. of Singapore, Singapore, Singapore)

Dolphins and porpoises use their sophisticated biosonar systems for targets detection, within a range of a few meters to about 200 m, there is not a better sonar on the planet. In this study, the high resolution computer tomography (CT) scan data were used to create the detecting click signal propagation models of Atlantic bottlenose dolphins (*Tursiops truncatus*) and harbor porpoise (*Phocoena phocoena*). The finite element methods (FEM) were used to simulate the processes of the clicks emitted from phonic lips and transmit to the water through animals' heads. The biosonar beam forming in the nearfield and farfield including the amplitude contours were determined and compared to the prior measurement results. There were no evidences of convergence in the farfield, which were consistent with measurement results for *Tursiops truncatus*. Additionally, in a cross-modal matching experiments with *Tursiops*, we found that the accuracy of the successive match was significantly different when the following subjects with same shape were used (water-filled PVC pipes, air-filled PVC pipes, foam ball array, and PVC pipes wrapped by foam) the results of FE model shed some light on the reasons why the animal has significant difference in performances when detecting different targets in the experiments.

2:15

2pAB6. Influences of spatial variability in pelagic scattering layers on sperm whales behavior. Adrienne M. Copeland (Univ. of Hawaii at Manoa, 1315 East West Hwy., SSMC3, Rm. 10246, Silver Spring, MD 96734, acopelan@hawaii.edu), Ladd Irvine, Bruce Mate (Marine Mammal Inst., Dept. of Fisheries and Wildlife, Oregon State Univ., Newport, OR), and Whitlow Au (Univ. of Hawaii at Manoa, Kailua, HI)

The central slope of the Gulf of Mexico (GOM) is home to over 20 species of marine mammals. Prior to the Deep Horizon Oil Spill in 2010, sperm whales were the predominately sighted large whale in the area. This study investigated the distribution and utilization of the north-central GOM by sperm whales two years after the spill by comparing sperm whale presence to micronekton distribution. Micronekton, organisms 2–20 cm, might be a key component in understanding sperm whale behavior as they are potentially an essential link in the food web between primary producers and higher trophic levels, including cephalopods—primary prey items of sperm whales. A Simrad EK60 echosounder operating at 38 kHz recorded the backscatter of micronekton throughout the northern GOM. The spatial distribution of micronekton backscatter was highly variable and may be driven by proximity to the Mississippi Delta. Sperm whales were sighted by trained visual observers and foraging was detected using a towed hydrophone array. The sperm whales were sighted most often in shallow slope waters less than 900 meters in areas with higher micronekton backscatter, suggesting that the waters above sloping bottoms with dense patches of mesopelagic micronekton support the prey of sperm whales.

2:30

2pAB7. Bottom characterization in the northern Gulf of Mexico using chirp sonar and ship noise data. Altan Turgut (Acoust., Naval Res. Lab., Naval Res. Lab., 4555 Overlook Ave., S.W., Washington, DC 20375, alturgut@yahoo.com), Warren Wood (Marine GeoSci., Naval Res. Lab., Stennis Space Ctr., MS), and Raymond J. Soukup (Office of Naval Res., Arlington, VA)

As a part of Prof. George Ioup's multi-institutional research initiative, chirp sonar bottom surveys were conducted near the Mississippi Canyon in the northern Gulf of Mexico during the Littoral Acoustic Demonstration Center (LADC) 2001 experiment. Bottom geoacoustic properties were estimated using a shallow-towed 2–12 kHz chirp sonar that provided sub-meter resolution and 60 m penetration. The inversion results showed that the experimental site is mainly covered with clayey sediment. In 2010, a second bottom characterization experiment was conducted at $\sim$25 miles NE of the LADC 2001 experiment site using a deep-towed chirp sonar. Deep-towed chirp sonar provided geoacoustic property estimations as well as low-grazing-angle bottom-loss measurements. As a source of opportunity, own-ship noise data were also used for geoacoustic parameter estimation and bottom-loss measurements. Geoacoustic parameter estimations and bottom loss measurements from the chirp-sonar and ship-noise data provided similar results that were also in agreement with those from the LADC 2001 experiment. [Work supported by ONR.]

2:45

2pAB8. Population densities of dolphins in the Northern Gulf of Mexico. Kendal Leftwich, Jack G. Labien, SydiCharise Austin, Evan Wellmeyer, and Juliette W. Ioup (Physics, Univ. of New Orleans, 1021 Sci. Bldg.,New Orleans, LA 70148, kmleftwi@uno.edu)

The Littoral Acoustic Demonstration Center–Gulf Ecological Monitoring and Modeling (LADC-GEMM) consortium collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015, returning to sites previously surveyed by LADC (2007, 2009, and 2010). Results presented here are produced using data recorded by the LADC-GEMM Environmental Acoustic Recording Systems (EARS) at the site closest to the BP oil spill, specifically for Rizzo, bottlenose, Clymene, and Pantropical-Spotted dolphins. Clustering and trained classifier techniques were used to isolate dolphin species. A population model developed by LADC-GEMM based on acoustic data collected from the EARS buoys is used to estimate the strength of delphinid recovery in the northern Gulf of Mexico after the BP oil spill. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

Session 2pBA

Biomedical Acoustics, Structural Acoustics and Vibration, 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

Chair's Introduction—1:00

Invited Papers

1:05

2pBA1. Quantitative analysis of the angiogenic microvasculature in tumor using multiple scattering and dual-frequency transducers. Aditya A. Joshi, Sibo Li (MAE, NCSU, 911 Oval Dr., Raleigh, NC 27695), Sunny Kasoji (BME, UNC, Chapel Hill, NC), xiaoning jiang (MAE, NCSU, Raleigh, NC), Paul Dayton (BME, UNC, Chapel Hill, NC), and Marie M. Muller (MAE, NCSU, Raleigh, NC, mmuller2@ncsu.edu)

The objective is to characterize angiogenic networks using contrast-enhanced multiple scattering. The methods relies on the measurement of the diffusion constant from the time evolution of the incoherent backscattered intensity. Dual-frequency arrays allow to emit a pulse at a frequency close to resonance frequency of microbubbles, and to receive the backscattered echoes at higher harmonics, resulting in better contrast-to-tissue ratio compared to single frequency transducers. At 8 MHz, we demonstrated that the diffusion constant enables the quantification of the vascular density and anisotropy in tumor-related vasculature in rat models *in vivo*. We show here that the results could be further improved using a dual-frequency approach, using two linear arrays with different central frequencies (7 MHz/18 MHz) and using a custom made dual frequency transducer (3 MHz/15 MHz). The minimum detectable distance between two vessel-mimicking cellulose tubes filled with contrast agents was measured via the diffusion constant. The dual-frequency approach drastically reduced the smallest detectable distance between tubes, which was found to be 25μm using a single frequency, 20μm using two linear arrays and 10μm using the custom dual-frequency array, respectively. These results show that measuring the diffusion constant with a dual frequency array can enable the quantification of vessel density with high accuracy

1:25

2pBA2. Ultrasound molecular imaging of the secreted tumor marker Netrin-1 in multiple breast cancer models. Jennifer Wischhusen (LabTAU U1032, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, 151 Cours Albert Thomas, Lyon 69003, France, jennifer.wischhusen@inserm.fr), Jean-Guy Delcros (Cancer Res. Ctr. Lyon, U1052, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Katheryne E. Wilson (Radiology, MIPS, School of Medicine, Stanford Univ., Stanford, CA), Benjamin Gibert (Cancer Res. Ctr. Lyon, U1052, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Rodolfo Molina (LabTAU U1032, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Patrick Mehlen (Cancer Res. Ctr. Lyon, U1052, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France), Juergen K. Willmann (Radiology, MIPS, School of Medicine, Stanford Univ., Stanford, CA), and Frederic PADILLA (LabTAU U1032, INSERM - French National Inst. of Health and Medical Res.; LabEx DEVweCAN of the Univ. of Lyon, Lyon, France)

With regard to biomarker expression, tumors are heterogeneous media displaying intra- and interpatient variability. Ultrasound molecular imaging (USMI) is a newly developed tool that can characterize heterogeneous tumors. Here, we focus on Netrin-1-overexpressing breast cancer. A newly developed targeted therapy interfering with Netrin-1 requires patient stratification. We developed Netrin-1-targeted microbubbles (MBs) and assessed USMI for identification of Netrin-1-positive breast cancer. *In vitro* flow chamber assays showed specific binding of anti-Netrin-1-MBs to human and murine breast cancer cell lines overexpressing Netrin-1. *In vivo*, in Netrin-1-positive transgenic or implanted mouse breast cancer models, USMI showed significantly increased signals with Netrin-1-targeted MBs, compared with isotype control MBs and targeted MBs after blocking of Netrin-1:—44.8%, 32.0%, 33.7% intensity for $MBs_{Netrin-1}$, $MBs_{Isotype}$, or after blocking, resp., in MMTV tumors;—28.2%, 13.7%, 13.4%, resp., in SKBR7 tumors;—60.9%, 47.8%, 47.9%, resp., in 4T1 tumors. In Netrin-1-negative tumors and in normal glands, targeted and control MBs showed no significant differences. Immunohistochemistry confirmed the expression of Netrin-1 in endothelial cells. In conclusion, USMI allowed the characterization of tumor heterogeneity and the differentiation between Netrin-1-positive and -negative tumors, and has the potential to become a companion diagnostic for breast cancer patient stratification.

Contributed Papers

1:45

2pBA3. Quantitative-ultrasound-based prostate-cancer imaging by means of a novel micro-ultrasound scanner. Daniel Rohrbach (Riverside Res. Inst., 156 William St., 9th Fl., New York City, NY 11215, drohrbach@RiversideResearch.org), Brian Wodlinger, Jerrold Wen (Exact Imaging, Markham, ON, Canada), Jonathan Mamou, and Ernest Feleppa (Riverside Res. Inst., New York, NY)

Currently, transrectal ultrasound (TRUS) guided biopsy is the only method for definitive diagnosis of prostate cancer. Our current study used a high-frequency (i.e., 29-MHz), transrectal, micro-ultrasound system (Exact-Vu^{TM} micro-ultrasound, Exact Imaging) to acquire RF data from 163 patients immediately before needle firing during 12-core biopsy examinations. Quantitative ultrasound (QUS) estimates of effective scatter diameter (ESD), effective acoustic concentration (EAC), midband (M), intercept (I), and slope (S) were calculated. Additional QUS estimates were derived including envelope statistics employing a Nakagami distribution and the envelope signal-to-noise ratio (ESNR). Estimate values were used to train linear-discriminant classifiers and performance was assessed using area-under-the-curve (AUC) values obtained from receiver operating characteristic (ROC) analyses based on 10-fold cross validation. A combination of ESS- and EAC-related parameters produced an AUC value of 0.75. When ESNR or PSA value was added as a feature, the AUC increased significantly to 0.77 or 0.78, respectively. The best classifier performance was obtained by combining envelope statistics, PSA, ESD, and EAC, which produced an AUC of 0.80. Our initial results with AUC values of 0.80 are very encouraging for developing a new tool for prostate-cancer biopsy guidance and treatment.

2:00

2pBA4. Anechoic lesion detection in a strongly scattering medium: Application to pulmonary nodules. Kaustav Mohanty and Marie M. Muller (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., College of Eng., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu)

We present an imaging methodology to detect the presence of an anechoic lesion in a highly scattering medium consisting of spherical air scatterers. Classical imaging methods have failed to image such media. In the method presented here, the incoherent backscattered intensity is extracted and the linear growth of the diffusive halo is tracked. Sudden changes in this growth indicates the presence of a target. This algorithm combined with the algorithm of Winslow *et al.* on contour detection enables us to predict the presence and the location of such a lesion. This method has been developed to detect the presence of pulmonary nodules and ground glass opacities in the lung parenchyma. Using a 128-element linear array transducer operating at 5 MHz, experimental results were obtained from a sponge phantom with an air volume fraction of 50%, in which gelatin nodules of diameter 5 mm and 8 mm were implanted at depths of 15 mm and 20 mm, respectively. The diameter of the nodules could be predicted within an error margin of $\pm$25%. FDTD simulations were also carried out with nodule sizes of 6mm and 11 mm, in media with 20% and 50% air volume fractions. The coordinates of the center of the lesion could be predicted within a tolerance of $\pm$7%.

2:15

2pBA5. Development of tissue-mimicking phantom of the brain for ultrasonic studies. Somayeh Taghizadeh, Cecille Labuda, and Joel Mobley (Dept. of Phys. and Astronomy, Univ. of MS, 123 Lewiss Hall, University, MS 38677, staghiza@go.olemiss.edu)

Constructing tissue-mimicking phantoms of the brain for ultrasonic studies is complicated by the low backscatter coefficient of brain tissue, causing difficulties in simultaneously matching the backscatter and attenuation properties. In this work, we report on the development of a polyvinyl alcohol (PVA) based tissue-mimicking phantom with properties approaching those of white matter tissue. PVA was selected as the base material for the phantom as its properties can be varied by temperature cycling, variations in concentration and the addition of scattering inclusions, allowing some independent control of backscatter and attenuation. The ultrasonic properties (including speed of sound, attenuation, and backscatter) were optimized using these three methods with talcum powder as a scatterer. It was determined that the ultrasonic properties of the phantom produced in this study are best matched to brain tissue in the frequency range 1.0–2.5 MHz, indicating its utility for benchtop ultrasonic studies in this frequency range.

2:30–2:45 Break

Invited Papers

2:45

2pBA6. Quantitative ultrasound for the dynamic biomechanical analysis of tongue-food interface during oral processing: An *in vitro* study. Mathieu Mantelet, Isabelle Souchon, Maud Panouillé, François Boué (UMR 782 GMPA, INRA - AgroParisTech - Université Paris Saclay, Unit GMPA, INRA-AgroParistech, 1 Ave. Lucien BrÕtigniÒres, Thiverval-Grignon 78850, France), Frédéric Restagno (UMR 8502 LPS, CNRS - Université Paris Sud - Université Paris Saclay, Orsay, France), and Vincent Mathieu (UMR 782 GMPA, INRA - AgroParisTech - Université Paris Saclay, Thiverval-Grignon, France, vincent.mathieu@inra.fr)

The development of non-invasive methods is critical for a better understanding of the biomechanical phenomena involved in the dynamic mechanisms of food texture perception during oral processing. The aim of the present study is to investigate *in vitro* the potential of a quantitative ultrasound device in order to monitor the mechanical properties of tongue-food interface during a compression. A tongue-palate bio-mimicking set-up was designed, consisting in a traction-compression machine equipped with tongue and palate phantoms. Gels and emulsion filled gels were considered as model foods for their wide ranges of texture properties. Finally, a 1 MHz ultrasonic transducer positioned under the tongue records in real-time the pulse-echo response of the tongue-food-palate system during a compression. The results put in evidence *in vitro* the potential of tongue-food interface reflexion coefficient to monitor the competitive and collaborative contributions (i) of the tongue-palate system (roughness, lubrication, and velocity) and (ii) of food properties (composition, structure, and rheology) on the mechanical properties of the tongue interface during occlusion cycles. The study paves the way for the use of quantitative ultrasound methods to monitor tongue-food-palate system during oral processing.

3:05

2pBA7. Ultrasound propagation in bone. 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)

Bone sonometry is a rapidly evolving diagnostic technology for managing osteoporosis. Bone sonometers that measure ultrasound propagation through calcaneus or along long bones (*parallel* to the long axis) have been used for many years. However, the Food and Drug Administration recently cleared two new, innovative designs that measure ultrasound propagation *perpendicular* to long axes of tibia (2016) and radius (2017). A basic understanding of wave propagation in bone is required in order to compare devices based on different physical principles and skeletal sites. Phantom studies elucidate the relative importance of components of signal loss during propagation through bone: absorption, longitudinal-to-shear scattering, and longitudinal-to-longitudinal scattering (Wear, *IEEE Trans. UFFC,* 55, 2418–2425, 2008). Comparison of experimentally-measured ultrasound properties (attenuation, sound speed, and backscatter coefficient) and finite-element-analysis-derived mechanical properties *in vitro* (n = 25) indicate that ultrasound measurements provide additional information regarding fracture risk beyond that provided by bone quantity alone (Wear *et al., Bone,* **103**, 93–101, 2017). Bone can support two longitudinal waves, which often overlap in time and frequency domains but can be separated using Bayesian probability theory or the modified least-squares Prony's (MLSP) method (Wear *et al.*, *J. Acoust. Soc. Am.*, **136**, 2015–2024, 2014).

3:25

2pBA8. A new dual-mode ultrasonic technique for assessing cortical bone. Jonathan J. Kaufman and Gangming Luo (CyberLogic, Inc., 611 Broadway, Ste. 707, New York, NY 10012, jjkaufman@cyberlogic.org)

Osteoporotic fractures are a major public health problem; diagnosis is based on x-ray densitometry (DXA); however, DXA is not able to accurately predict who will and will not suffer a fracture. An alternative to DXA is ultrasound, which is viewed as having the potential to better characterize fracture risk. Our approach is based on "dual-mode" ultrasound in which 2 distinct modes are used together to assess cortical bone. In particular, measurements are made at the mid-shaft tibia in both axial-transmission and pulse-echo modes. Axial transit time (τ_A) and pulse-echo transit time ($\tau_{P/E}$) are obtained at the same tibial cortical site. These 2 transit times can be used in a classification scheme to identify individuals at greatest risk of fragility fracture. A pilot set of measurements on 5 individuals has been carried out to demonstrate the basic feasibility of the proposed technology. For the 5 subjects, each subject data point was found to be located in a distinct region (i.e., a hypothesized distinct bone quality state) of the quadrant associated with the 2 transit times. Because the 2 transit times are each affected in distinct fashions by a variety of bone quality factors (such as degree of mineralization, porosity, cortical thickness, and biomechanical stiffness), the bivariate feature has the potential to accurately identify those individuals at increased risk of fracture. Future studies on individuals with and without fragility fractures will elucidate the capabilities and potential of the dual-mode ultrasound technology.

Contributed Paper

3:45

2pBA9. Diffusion constant and frequency dependent attenuation for the assessment of cortical bone porosity. Omid Yousefian, Yasamin Karbalaeisadegh (Mech. and Aerosp. Eng., North Carolina State Univ., 2704 Brigadoon Dr., Apt. A, Raleigh, NC 27606, ykarbal@ncsu.edu), Kay Raum (Charité - Universitätsmedizin Berlin, Berlin, Germany), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

The objective of this study is to assess cortical bone porosity through measurements of the diffusion constant and attenuation as a function of frequency. Using FDTD simulations, the diffusion constant was calculated for 45 modified human femur shaft bone geometries obtained from a reference binarized 100-MHz Scanning Acoustic Microscopy image. The diffusion constant was sensitive to cortical porosity and demonstrated around 85% decrease for 10% increase in porosity. This suggests that lower diffusion constant values could be associated with more severe cases of osteoporosis. Using a spectroscopy approach in the 1–8 MHz range, we demonstrate that the attenuation coefficient as a function of frequency can be approximated as:$\alpha(f) = Af^B + C$, where "A", "B" and "C" are related to pore diameter and pore concentration. For a constant pore density (10 mm^2), "A" increased from 0.03 to 19.79 when pore diameter increased (Po.Dm $\in$40–130μm), while "B" and "C" were found to decrease from 2.654 to 0.5741 and from −0.061 to −21.43, respectively. A similar trend was observed when pore diameter was kept constant. A model was derived to quantify pore size and concentration through constrained nonlinear optimization method. This parametric model combined with estimation of the diffusion constant has the potential to enable the quantification of pore size and concentration.

Invited Papers

4:00

2pBA10. *In vivo* radius bone evaluation in their teens by two longitudinal wave propagation. Mami Matsukawa (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Isao Mano (Doshisha Univ., Joyo, Japan), Yutaro Yoneda, Kaoru Horii (OYO Electric Co., Joyo, Japan), Shiori Umemura, and Etsuko Ozaki (Kyoto Prefectural Univ. of Medicine, Kyoto, Japan)

An ultrasonic bone measurement system, LD-100 (OYO electric),[1] has been developed for the evaluation of the distal 5.5% site of the radius of the non-dominant hand. The system measures two longitudinal waves (fast and slow waves) which propagate in the inside cancellous bone and echo waves reflected from the interface between the cortical and cancellous bones. The properties of these waves can give us the information of cancellous bones. We can also estimate the cortical thickness from the echo waves. For the small radius of teenagers, we have improved the system using an annular type transducer to avoid the effects of guided waves in the cortical bone. The radius bones of 654 teenagers were measured. The cortical thicknesses of female students in their late teens were around 95 % of the young adult mean (YAM), where those of male students showed variation from 90 to 100%. The cancellous bone densities in their late teens were 82–94% (female) and 66–85% (male). The growth of cancellous bone was late, which was clearer in men. In addition, the total bone growth of men seemed slower than that of women. [1]H. Sai, *et al*., Osteoporos Int. (2010) **21**:1781.

4:20

2pBA11. Associations between ultrasonic backscatter, bone density, and microstructure in cancellous bone characterization. Dean Ta (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn) and Chengcheng Liu (Inst. of Acoust., Tongji Univ., Shanghai, China)

Ultrasonic backscatter is related to the microstructure of cancellous bone. However, whether backscatter measurement can provide additional information independently of BMD remains speculative. In this study, we analyzed the independent contribution of cancellous bone densities and microstructure to apparent backscatter signals *in vitro*. The results showed that ultrasonic backscatter and the statistics of backscatter envelope were significantly correlated with bone densities and microstructure. After adjustment for BMD, some trabecular structures still contributed significantly to the adjusted correlations, with moderate additional variance explained. Multiple linear regressions revealed that both bone density and structure contributed significantly to the prediction of ultrasonic backscatter (adjusted $R^2 = 0.75–0.93$, $p<0.05$), explaining an additional 14.0% of the variance at most, compared with that of BMD measurements alone. We also demonstrated that the Nakagami model had great potential in trabecular microstructure characterization. The results proved that ultrasonic backscatter was primarily determined by bone apparent density, but bone densities plus microstructure structure could achieve encouragingly better predictive performance than BMD alone. This study implied that ultrasonic apparent backscatter might provide additional density and structural features unrelated to current BMD measurement. Thus, we suggest that ultrasonic backscatter measurement could play a more important role in cancellous bone evaluation.

Contributed Paper

4:40

2pBA12. Assessment of dental implant stability: Coupling modeling and experimental approaches. Romain Vayron, Vu-Hieu Nguyen, 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 clinically but implant failures, which may have dramatic consequences, still occur and remain difficult to anticipate. Accurate measurements of implants biomechanical stability are of interest since they could be used to improve the surgical strategy. The aim of this work is to develop a medical device capable of estimating dental implant stability using quantitative ultrasound. To do so, our approach consists in coupling modeling and experimental methods. Firstly, an implant was 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 is derived. Then, the implant is unscrewed by 2π radians and the measurement was realized again. The value of indicator significantly increases as a function of the number of rotation. The results show that bone quantity in contact with the implant has a significant influence on its ultrasonic response. Secondly, a 3D finite element model is employed and the geometrical configuration is assumed to be axisymmetric. The numerical results show that the implant ultrasonic response changes significantly when a liquid layer is located at the implant interface.

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics V

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

Contributed Papers

1:00

2pEA1. Timbre analysis from monotone rearranging of the acoustic signal: Application to fountains. Laura Velardi (LISA-Environ. HydroAcoust. Lab, ULB, av. F.D. Roosevelt 50, Brussels 1050, Belgium, lvelardi@ulb.ac.be), Jean-Pierre Hermand (LISA-Environ. HydroAcoust. Lab, ULB, Brussels, Brussels Capital, Belgium), and D'Autilia Roberto (Matematica, Università degli Studi di Roma Tre, Rome, Italy)

Public fountains are an example of a soundmark that is both non musical and pleasing to the ear due to their specific timbre. Standard audio descriptors designed for musical sounds are limited in describing such sounds. A new method is proposed for identifying and classifying fountains sounds as well as surrounding urban noises. The method consists in sorting discrete-time signal samples in a monotonically descending order. Features of the resulting curve are extracted in an attempt to define a timbral evaluation metric. Reference signals including natural water sounds and synthesized pink noise are also analyzed and the results are compared to those of fountain sounds. Outcomes are discussed in the framework of urban soundscape studies, in order to improve acoustical urban planning.

1:15

2pEA2. Assessment of granular distortion in digital sound recorders by statistical analyses of recorded sound and analytically distinguishing quality of 24-bit and 16-bit analog to digital converters. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Modern sound recording systems are mostly digital type, using at least one or several analog to digital converters that commonly provide 24-bit or 16-bit resolution and accordingly limits the recording quality. The use of such digitizers inherently add some granulation distortion. A 16 bit digitizer is theoretically limited to granulation step size of 1/65,536. Whereas a 24 bit digitizer is theoretically limited to granulation step size of 1/16,777,216. Smaller granulation step size provides better fidelity of the recording. This study explored the operational resolution of a few of such digital recorders by statistical analysis of samples of recorded sound tracks. The histogram of the digital samples of the sound recordings showed that a digitizer may not always be digitizing to the finest precision.

1:30

2pEA3. A micromachined microphone based on the membrane with bias voltage and field-effect-transistor mechano-electrical transduction. Junsoo Kim, Hoontaek Lee, Donghwan Seo, Jaehyeok Jin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), San 31 Hyoja-dong, Nam-gu, Pohang, Gyeongsangbuk-do 37666, South Korea, chak7258@postech.ac.kr), Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyungbuk, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

Recently, capacitive-type microphone dominates the MEMS (Micro-Electro-Mechanical-System) microphone market. Since sizes of the electrodes on the membrane and backplate determine the sensitivity of the microphone, there is a limit in reducing the size of the MEMS microphone. Here, we propose a micro-machined microphone consist of the membrane with bias voltage and field-effect-transistor (FET). The difference between the conventional capacitive MEMS microphone and proposed microphone is the Mechano-Electrical transduction mechanism. By biasing the voltage on membrane, the gate voltage is induced depending on the vibration of the membrane, eventually changes the source drain current. In case of a MEMS microphone using the proposed FET and biased membrane microphone, low frequency roll-off according to the energy conversion does not occur, and the size of the membrane and backplate can be reduced as compared with the conventional MEMS microphone, without the degradation of sensitivity. In addition, depending on the biasing condition, amplitude modulation technique to reduce the flicker noise of the FET can be applied, which leads to the higher SNR (Signal to Noise Ratio). Conventional metal-oxide-semiconductor fabrication process and micromachining process were used. In this study, design, fabrication and performance test of the proposed FET microphone are conducted. [Work supported by CMTC, UM15304RD3.]

1:45

2pEA4. A micromachined microphone based on the electret membrane and field-effect-transistor mechano-electrical transduction. Junsoo Kim, Hoontaek Lee, Donghwan Seo, Jaehyeok Jin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyeongsangbuk-do, South Korea), Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Gyungbuk, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang city, Kyungbuk, South Korea, Pohang, Kyungbuk 37666, South Korea, wkmoon@postech.ac.kr)

Most commercially available MEMS (Micro-Electro-Mechanical-System) microphones use a capacitive method, and their structure and performance are saturated to some extent. However, due to the limitation of the capacitive transduction method, the roll-off at the low frequency is inevitable, and there is a limit in reducing the mechanical thermal noise caused by the squeeze film damping that occurs between membrane and backplate structure. Proposed electret membrane and FET (Field Effect Transistor) based MEMS microphone detects a change in the source-drain current according to the gate voltage change of the FET induced by vibrating the membrane with the fixed charges. In the case of a MEMS microphone using the proposed FET, low frequency roll-off according to the energy conversion does not occur, and the size of the backplate can be drastically reduced as compared with the conventional MEMS microphone, thereby further reducing the mechanical thermal noise, leading to the possibility of achieving the higher SNR (Signal to Noise Ratio) than 65 dBA. Conventional metal-oxide-semiconductor fabrication process and micromachining process were used. In this study, design, fabrication and performance test of the proposed FET based MEMS microphone are conducted. [Work supported by CMTC, UM15304RD3.]

TUESDAY AFTERNOON, 5 DECEMBER 2017 BALCONY I/J/K, 2:00 P.M. TO 3:00 P.M.

Session 2pEDa

Education in Acoustics: Acoustics Education Prize Lecture

Chair's Introduction—2:00

Invited Paper

2:05

2pEDa1. Student-centered acoustical engineering education at the University of Hartford. Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

The University of Hartford has provided the author with a fulfilling métier, shepherding two undergraduate engineering programs in the area of acoustics: (1) the Bachelor of Science in Mechanical Engineering (BSME) with Acoustics Concentration and (2) the Bachelor of Science in Engineering with a major in Acoustical Engineering & Music. The first is part of our BSME degree program that has required courses in Vibration as well as Engineering Acoustics since the 1960's. The Acoustical Engineering & Music degree is a unique program instituted in 1976, where applicants must meet engineering's math and science entrance requirements as well as pass the audition requirements of our music conservatory (The Hartt School). Both ABET-accredited programs encompass the same engineering vibrations and acoustics courses, as well as the same acoustics projects sequence, beginning in the sophomore year. Alumni of both undergraduate programs have successfully obtained positions in consulting (architectural and environmental), audio product and A/V design, musical instrument design, hearing- and psychoacoustic-related design, noise/vibration control of machines, as well as graduate degrees. The use of real world industry-sponsored acoustic projects for engineering design courses throughout the curriculum will be described, as well as the programs' Service Learning components and High Impact Practices.

TUESDAY AFTERNOON, 5 DECEMBER 2017 BALCONY I/J/K, 3:15 P.M. TO 4:15 P.M.

Session 2pEDb

Education in Acoustics: General Topics in Acoustics Education

Contributed Papers

3:15

2pEDb1. Writing an acoustics textbook under the ASA Press/Springer contract. Steven L. Garrett (None, Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

The Acoustical Society's Books+ Committee decided to add first editions to their "classic reprints." After a search process, the ASA decided that it was in the best interests of that project to team with Springer. A contract between ASA and Springer was signed in May 2013 creating a mechanism by which ASA members could propose and write new books. Members, student members, or registered meeting attendees could purchase paperback versions of those books produced under that contract for $24.99. Having just published my acoustics textbook [*Understanding Acoustics*; ISBN 978-3-319-49976-5], I will report on my experiences during the entire process, which begins with a book proposal. If the Committee believes that the proposed book would serve the interests of the membership, an individual contract is signed. Springer provides a "template" in MSWord^{â} or L^{a}T$_{E}$X format. I used the Word version that operated like any *.docx. Once the manuscript was complete, two ASA members were assigned as content editors for the technical review. Upon their approval, the layout, some redrawing of artwork, and a "grammar and spelling" review were done very quickly by Springer's affiliate in India. I corrected my "author proofs" at a Springer web site that was amazingly user friendly.

3:30

2pEDb2. Development of a simple replica of the middle ear pathology for demonstration of tympanometric measurements. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

This study evaluates a simple replica of the middle ear pathology for demonstration of tympanometric measurements. Common tympanometers often use small passive cavities for calibration of basic measurements such as equivalent ear canal volume. But the real ear normally presents an inverted V shaped tympanogram. The pathologic ears present various characteristic shapes of tympanograms such as unusually high or low peaks. In some patients the tympanometric peak may be shifted away from the zero differential pressure point. Often such patients may not be readily available during a classroom demonstration of the procedures. Utility of this study to replicate such pathologic conditions with a simple model will be discussed.

3:45

2pEDb3. Engaging high school students in studying marine mammals observed near the BP oil spill. Kendal Leftwich, Juliette W. Ioup, C. Gregory Seab, Simeon P. Benit, and Matthew Firneno (Physics, Univ. of New Orleans, 1021 Sci. Bldg., New Orleans, LA 70148, kmleftwi@uno.edu)

The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project partnered with Warren Easton Charter High School to analyze underwater acoustic data collected in the northern Gulf of Mexico during the summer of 2015, returning to sites previously surveyed by LADC. Results presented here are produced using data recorded by Environmental Acoustic Recording Systems (EARS) at the site closest to the BP oil spill. The aim of the outreach project described here is for high school students to learn to work independently using scholarly works to analyze and interpret data, as well as to communicate and collaborate in a professional environment. Various marine mammals have been detected, including sperm whales, beaked whales (Cuvier, Blainville, Gervais, and BWG), and dolphins (bottlenose, Rizzo). The numbers and types of marine mammals are possible indicators of the health of the area and the marine mammals around the oil spill. Changes will indicate possible future problems. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

4:00

2pEDb4. Educational practice of linear array techniques at the Acoustics department of the Escuela Superior de Ingeniería Mecánica y Eléctrica campus Zacatenco, Instituto Politécnico Nacional, Mexico. Eduardo Vivas (CIBNOR, La Paz, Baja California Sur, Mexico), Omar A. Bustamante (Acoust., ESIME, IPN, Cuautitlan Izcalli, Mexico), and Sergio Beristain (Acoust., ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Sound Source Location is a topic that has been studied over the past decades for a wide range of applications whether in air or underwater, resulting in the development of different sophisticated methods and systems. Nevertheless, this topic in not usually included in the curricula of acoustic courses for the requirement of expensive and specialized instrumentation. The purpose of this research is to develop a practical approach looking to involve the students by using common hardware for which they are already familiar with: hardware from the videogames industry. In the Acoustics department of the Escuela Superior de Ingeniería Mecánica y Eléctrica (ESIME Zacatenco), IPN Mexico, a low-cost Sound Source Location system has been implemented for this purpose. The system is built around a PS3 Eye digital camera that includes a linear array of four microphones, and Raspberry Pi microcomputer. The Beamforming algorithms are programmed in Python to provide flexibility and ease of experimentation. It was originally designed for research applications, and now it is a prototype for laboratory practice. The system provides the undergraduate students with the basics of linear arrays and associated processing techniques engaging the students by using familiar hardware from the games industry and real life applications.

TUESDAY AFTERNOON, 5 DECEMBER 2017 STUDIO 7, 2:00 P.M. TO 3:30 P.M.

Session 2pID

Interdisciplinary: Guidance from the Experts: Applying for Grants and Fellowships

Martin S. Lawless, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Michaela Warnecke, Cochair
Dept. of Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St., Baltimore, MD 21218

Matthew C. Zeh, Cochair
Mechanical Engineering Graduate Acoustics Program, University of Texas at Austin, 1300 West 24th Street, Apt. 212, Austin, TX 78705

A panel of successful fellowship winners, selection committee members, and fellowship agency members will answer questions regarding grants and fellowships, application advice, and funding opportunities. The panelists will briefly introduce themselves, followed by a question and answer session with the audience.

Session 2pMUa

Musical Acoustics: Cajun Music: Accordians, Culture, and History

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

Chair's Introduction—1:55

Invited Papers

2:00

2pMUa1. Early history of the accordion family: Where did all the accordions come from? James P. Cottingham (Physics, Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

Two hundred years ago, there were no accordions. Free reed instruments were known in Asia for thousands of years, but the free reed instruments of European origin such as the accordion, harmonica, 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 that 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, in particular the accordion-concertina family, along with discussion of their acoustical design characteristics. Also addressed is the question of the influence of the Asian free reed mouth organs on the origin of the Western free reeds.

2:20

2pMUa2. Fluid dynamics approach to free reed physics. Thomas Tonon (Bluesbox, 11 Bolfmar Ave., Princeton Junction, NJ 08550, tsbtonon@gmail.com)

The periodic vibration of a free reed subject to a steady pressure gradient (bellows pressure) is examined from a fluid dynamics approach. In this analytic study, the self-excited forcing function is modeled from the static and dynamic pressures acting on the vibrating tongue surface. A first formulation includes standard mounting of the reed over a cavity, with a hole that accommodates a mean air flow to the outside. A second formulation includes in addition a resonant air geometry that could include non-standard operation, such as pitch bending.

2:40

2pMUa3. Cultural significance of the diatonic single-row button accordion in South Louisiana. Mark F. DeWitt (School of Music and Performing Arts, Univ. of Louisiana at Lafayette, UL Lafayette School of Music, PO Box 43572, Lafayette, LA 70504-3572, dewitt@louisiana.edu)

Musical acoustics are intimately bound up in culture. Musical instruments are vehicles for artistic expression in terms of visual design, timbre, and musical style for musicians who play them and sometimes for the artisans who make them. In complex multicultural societies, certain instruments can also become icons for group identity, similarly marked in visual, timbral, and musical terms. The case of the melodeon, known in Louisiana as the single-row button accordion or the Cajun accordion, richly exemplifies these possibilities. The instrument arrived in Louisiana in the mid-to-late 1800s and became the instrument of choice to play at house dances and dance halls by the 1920s, adopted by two neighboring ethnic groups, French-speaking Creoles of color and Cajuns. Local artisans began making single-row accordions when the supply from Germany ceased during World War II, creating a new, higher-quality version of the instrument with a distinctive appearance that continues to be the instrument of choice for Cajun accordionists and some zydeco players. This paper will elaborate on the history of the melodeon in Louisiana, the musical styles that use it, and its status as a regional and ethnic identity symbol.

3:00

2pMUa4. The craft of building Cajun accordions. Larry G. Miller (Bon Cajun Accordions, 886 McMillan Ave., Iota, LA 70543, boncajun@charter.net)

Craftsmen have been building single-row diatonic button accordions in South Louisiana since the mid-twentieth century. A brief history of this phenomenon will be given, followed by an explanation of how these accordions are constructed, from the perspective of someone who has made these instruments for decades, trained other makers, and served as the parts supplier for several shops. Topics covered will include the types of wood used, other parts and materials, design choices that affect sound quality versus visual appeal, special challenges in assembly, and tunings.

3:20

2pMUa5. Cajun and Zydeco accordion playing styles. Chad Huval (Univ. of Louisiana at Lafayette, 239 Washington St., Breaux Bridge, LA 70517, cchuval@yahoo.com)

The 10-button single-row diatonic accordion is a mainstay in the music of Louisiana's Creoles of Color and Cajuns. From the single-row diatonic accordion's appearance in the earliest recordings of Louisiana music to now, a unique syncopated playing style has developed, leading to today's zydeco and Cajun styles. This talk will focus on how the 10-button single-row diatonic accordion is played, with demonstrations of the nuances that make it Cajun, or make it zydeco.

3:40–4:05 Panel Discussion

TUESDAY AFTERNOON, 5 DECEMBER 2017 SALON E, 5:00 P.M. TO 6:00 P.M.

Session 2pMUb

Musical Acoustics: Cajun Music Concert by the Savoy Family Band

James P. Cottingham, Chair

Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

The Savoy Family Cajun Band plays honed down, hard-core Cajun music laced with sounds of fiddle and accordion rooted in South Louisianan tradition. Marc and Ann Savoy and their sons, Joel and Wilson, work together to create a tight, intense sound. Marc Savoy is widely recognized as a player and builder of the Cajun accordion and has received the National Heritage Fellowship Award. Marc and Ann have been performing all over the world and recording together since 1977. They appeared on the PBS series "American Roots," for which Ann wrote the chapter on Cajun music in the book that accompanied the series. Joel and Wilson are each distinguished musicians, both a soloists and members of other groups. The Savoy Family Band brings the energy of the dancehalls of southwest Louisiana to the stage, peppered with informative anecdotes about life in the Cajun heartland.

TUESDAY AFTERNOON, 5 DECEMBER 2017 STUDIO 2, 1:00 P.M. TO 2:55 P.M.

Session 2pNSa

Noise, Architectural Acoustics, and ASA Committee on Standards: Evaluation of Acoustics in Hospitals and Healthcare Facilities

Jay Bliefnick, Cochair

Architectural Engineering & Construction, University of Nebraska, 1110 S 67th St., Omaha, NE 68182-0816

Jonathan R. Weber, Cochair

Durham School of Architectural Engineering & Construction, University of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Evaluating hospital soundscapes to improve patient experience. Jay Bliefnick, Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, jbliefnick@huskers.unl.edu), and Rebecca Jackson (Nebraska Medicine, Omaha, NE)

Patients routinely perceive hospital soundscapes to be poor when rating their experience on HCAHPS (Hospital Consumer Assessment of Healthcare Providers and Systems) surveys administered after discharge. In this study, sound levels within five hospital units were correlated with HCAHPS noise perception survey data. Acoustic metrics including A-weighted equivalent, minimum, and maximum (LAEQ, LAMIN, & LAMAX) and C-weighted peak (LCPEAK) sound pressure levels, occurrence rate, and speech intelligibility index were evaluated in 15 patient rooms and 5 nursing stations. Average patient room LAEQ values within the five units ranged between 52 dBA and 61 dBA with speech intelligibility ranging from poor (<0.45) to marginal (0.45 to 0.75). The absolute minimum values measured within the patient rooms (LAMIN, LCMIN, & LZMIN) were found to be correlated with HCAHPS data and other metrics revealed trends consistent with patient perception. For example, the lowest rated unit also had higher occurrence rates, indicating this unit was louder more often. Ceiling type was also found to impact sound levels with LAEQ 5 dBA quieter on average for rooms utilizing acoustic tile ceilings. Taken as a whole, these results provide insight into acoustic metrics and design strategies which can ultimately be utilized to improve patient experience.

1:25

2pNSa2. Acoustical evaluation of quiet time and its impact on patient outcomes. Jonathan R. 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), Myra Rolfes, Heather Cooper (Neonatal Intensive Care Unit, Children's Healthcare of Atlanta, Atlanta, GA), Brooke Cherven (Nursing Res. & Evidence Based Practice, Children's Healthcare of Atlanta, Atlanta, GA), and Ashley Darcy Mahoney (The George Washington University School of Nursing, Autism and Neurodevelopmental Inst., Washington, DC)

As healthcare evolves from focusing on survival to prioritizing patient care, more efforts have been devoted to exploring the characteristics of an optimal environment. Intensive care units (ICUs) often involve more urgent situations that require more medical equipment, in-unit procedures, and staff that all contribute to the noisy soundscape. Recently, administrative noise reduction strategies such as Quiet Time (QT) and behavioral modification and noise awareness programs have gained popularity. The literature typically demonstrates some positive effects from these interventions; however, there are inconsistencies in magnitude and sustainability. We conducted an 18-month, longitudinal study of QT aimed at (a) characterizing the Neonatal Intensive Care Unit (NICU) soundscape more extensively and with newly developed metrics and (b) determining the impacts of sound on patient physiological responses. Results show some evidence of trends toward decreased noise levels after QT implementation. For example, QT generally resulted in lowered occurrence rates, indicating a decreased number of high-level noise events. Additionally, spectral data showed significant decreases in level across the vocal frequency range and improved speech intelligibility index scores with QT implementation. Ultimately, these results begin to provide a template for determining the optimal soundscape to aid more targeted administrative and architectural design strategies.

1:45

2pNSa3. An overview of the key misperceptions that are preventing any significant improvements when addressing operational noise in acute care hospitals. Gary Madaras (Making Hospitals Quiet, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

The Making Hospitals Quiet program has been working with healthcare providers inside their existing and noisy acute care hospitals since 2011. An overview will be given on why misperceptions by healthcare professionals and acousticians have resulted in little improvement in the collective patient perception of quietness while admitted in these facilities. Tactics for overcoming these misperceptions will be discussed.

2:05

2pNSa4. Acoustics and the built environment in pediatric hospital units. Ian R. Hough, Yoshimi Hasegawa, and Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, ian.r.hough@gmail.com)

Previous studies have indicated that noise in hospitals can present challenges for patients and staff such as sleep disruption and increased stress symptoms. However, few studies have looked more holistically at the built environment to include evaluations of aspects in addition to noise, such as lighting, thermal comfort, and air quality. In this study, built environment measures and staff perception surveys were conducted in two units: a pediatric intensive care unit (PICU) and a pediatric medical-surgical unit (MedSurg). Results were evaluated to understand physical and perceptual differences based on the time of day, type of unit, and location within the unit. Results revealed interesting trends about acoustics and the built environment. For example, perception of noise, lighting, thermal comfort, and odor differed depending on the location within the unit (nurse stations, patient rooms, and corridors). Many perceptual results tracked with trends in the built environment measures. For example, day shift workers were significantly more annoyed with noise. Likewise, acoustic occurrence rates were generally found to be higher during the day, indicating the units were louder more often in the daytime with shorter periods of restoration. Viewed holistically, these findings reveal potential opportunities to improve the overall built environment in hospitals.

Contributed Papers

2:25

2pNSa5. Noise in the recovery area of a large hospital. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A noise study was made in a very large government hospital which includes eight surgery rooms and a very large recovery area where patients are placed from as little as half an hour, either before or after a surgery, to about ten hours after some given surgery. In this area, physicians, nurses and bed moving personnel activity is almost continuous, and it has plenty of "Life Equipment" monitoring each and every patient. The noise measurements were concentrated in the recovery area, which is a very large room with very hard boundaries, that includes over 25 beds and aisle areas with no more than curtains separating them in order to allow for easy movement of rolling beds in and out from the surgery area and to the different specialty floors of the hospital. Results of these measurements are presented and commented.

2:40

2pNSa6. A study on 7 color marking of impact noise. Uk-Jin Song and Myungjin Bae (Dept. Information and TeleCommun., Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156743, South Korea, imduj@ssu.ac.kr)

Interlayer noise is a noise pollution that occurs mainly in a space where a large number of households, such as multi-family houses or apartments, are together. There are various types of interlayer noise depending on what kind of sound is generated, but there are impact sounds as a great stimulus to human auditory sense. However, to date, the issue of interlayer noise has been the subject of concern because the interlayer noise criterion limits the average size of a certain section and there is no standard for impact noise that appears temporarily. Therefore, in this paper, A7B shows the number of impulse sounds in a row in order to make objective visual data about the interlaminar impact noise. In this case, the criterion of the impact sound is defined as a portion where a large change in sound volume occurs within 0.1 second. Experimental results show that the results similar to the number of impact sounds of the corresponding interlaminar impact noise source can be visually confirmed, and furthermore, it is possible to present clear and objective data in the interstory noise dispute.

TUESDAY AFTERNOON, 5 DECEMBER 2017 STUDIO 2, 3:30 P.M. TO 5:10 P.M.

Session 2pNSb

Noise, Architectural Acoustics, Speech Communication, Psychological and Physiological Acoustics, and ASA Committee on Standards: Acoustics and Its Role in Accessibility (e.g., Americans with Disabilities Act)

David Manley, Chair
D.L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218

Chair's Introduction—3:30

Invited Papers

3:35

2pNSb1. Disability rights aspects of ambient noise under the Americans with Disabilities Act for people with auditory disorders. Daniel Fink (The Quiet Coalition, The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

The goals of the disability rights movement are full enjoyment, active inclusion, and equal participation in society for people with disabilities. The Americans with Disabilities Act (ADA) guarantees full and equal enjoyment of places of public accommodation for them. People with partial hearing loss, tinnitus, and hyperacusis meet the ADA definition of having a disability but currently have no legal protection under ADA. People with 25–40 decibel hearing loss cannot understand speech in noisy places, with or without hearing aids. They require ambient noise levels below 60 A-weighted decibels with reverberation times under 0.50 seconds to understand speech. Noise worses symptoms for those with tinnitus and hyperacusis. Noisy restaurants and other indoor places deny full enjoyment and equal participation in public life to people with these auditory disorders. Legislative and regulatory actions are needed to provide quiet environments, with established noise standards vigorously enforced. Technologies and environmental modifications to control noise and reverberation are well known, readily available, and relative inexpensive. The simplest modification is merely turning down the volume of amplified sound. Universal design for quiet facilitates communication for everyone and prevents hearing loss, tinnitus, and hyperacusis in those without auditory disorders.

3:55

2pNSb2. Preliminary reports of New York city restaurant and bar noise levels via crowdsourced measurements show a low number of restaurants and bars suitable for various disabled communities, notably those with hearing loss. Gregory Farber (SoundPrint, PO Box 533, Lincoln, MA 01773, greg@soundprint.co)

Noise causes hearing loss, tinnitus, and hyperacusis, and interferes with understanding speech for those with normal hearing and for those with hearing loss. Media reports document high sound levels in restaurants and bars, but accurate sound measurements are lacking. Noise pollution is an important health and social issue and the public needs a reliable way to determine whether certain venues are safe for auditory health and conducive to conversation. This is important for the disabled, the blind, the autistic and the hearing impaired, who include a large number of older Americans. This report is of sound level measurements in more than 2250 venues in New York City, using the novel SoundPrint smartphone app. The average sound level was 78 dBA in restaurants and 81 dBA in bars. These sound levels do not allow ready conversation, even for those with normal hearing and certainly not for those with hearing loss. There were variations in sound levels in restaurants in different neighborhoods and in different types of restaurants. The reported sound levels by venue managers generally underestimated actual sound levels. This report is a proof-of-concept study of crowdsourced sound measurements, which can provide valuable data for the general public and health officials.

4:15

2pNSb3. Electric vehicles—Safe and quiet with acoustic vehicle alerting systems? Rene Weinandy (Noise Abatement in Transport, German Environment Agency, Woerlitzer Platz 1, Dessau-Rosslau 06844, Germany, rene.weinandy@uba.de), Lars Schade, and Jan Gebhardt (Noise Abatement in Transport, German Environment Agency, Dessau-Rosslau, Saxony-Anhalt, Germany)

One of the most important environmental issues in densely populated areas is the problem of noise. Traffic noise from cars, railway vehicles and airports located in close proximity to cities is not only annoying for residents; it also leads to serious health issues and has an enormous negative economic impact. Due to this, it is of primary importance to make our cities quieter. The German Environment Agency is working on noise and its effects on humans, especially with respect to policy and regulation. Currently there is a strong debate here in Germany—as in many other parts of the world—about road vehicles that rely, in whole or in part, on alternative drive trains. On the one hand, there are positive environmental benefits expected from these "hybrid or pure electric" road vehicles. On the other hand, these vehicles are thought to pose a risk to blind and low vision pedestrians, due to their lower noise emissions at approach. To address this concern, the European Union has legislated that future hybrid and pure electric cars must be equipped with acoustic vehicle alerting systems (AVAS). The presentation provides a critical assessment of the effectiveness of AVAS and of their negative side effects.

4:35

2pNSb4. Induction loop assisted listening system case study in a Catholic church. David Manley and Benjamin Bridgewater (D.L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, dmanley@dlaa.com)

A Catholic church in Metro Denver, Colorado area, was undergoing major renovations for the first time since the 1970s, and wished to incorporate an induction loop assisted listening system as an amenity for their aging members. The pastor indicated the Church has lost a number of parishioners due to the lack of an assisted listening system. While the Church installed a temporary radio frequency assisted listening system until renovations could be complete, a permanent and integrated system was desired. The case study presents coordination with Church, the loop system manufacturer, testing of the existing space for metal loss, and design results.

Contributed Paper

4:55

2pNSb5. On identifying the accidental sudden acceleration of an SUV vehicle by engine sound. Jihye Bae and Myungjin Bae (Commun. Eng., Soongsil Univ., 369 sangdoRo, DongjakGu, Seoul 156-743, South Korea, jbae5@ssu.ac.kr)

A major accident occurred when an SUV vehicle of the company A that recently ran the road was flooded with roadside trees on the road. It was suspected of a sudden accident, but Company A judged the driver immature because the driver did not step on the brake paddle at that moment. The situation before and after the accident was recorded in the black box video, but the engine sound was not heard well. However, we analyzed the curve of the change in engine sound while emphasizing the change of the engine sound at the moment of the accident by recording the voice in the black box video. In particular, when comparing the curve of the engine sound change below 1,000 Hz with the rapid acceleration curve of the existing engine sound, for 5 seconds before the accident, the rotation sound of the engine changed 1.7 times faster than the rapid acceleration, and the highest rpm lasted 3 seconds. After all, in terms of the engine sound change curve of the black box video, this was a sudden velocity accident. In this paper, we proposed a new method to identify the accident through engine sound.

TUESDAY AFTERNOON, 5 DECEMBER 2017 BALCONY L, 1:00 P.M. TO 3:00 P.M.

Session 2pPA

Physical Acoustics: General Topics in Physical Acoustics I

Kyle S. Spratt, Chair
Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713

Contributed Papers

1:00

2pPA1. Champagne bubble acoustics. Kyle S. Spratt, Kevin M. Lee, and Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, sprattkyle@gmail.com)

Sparkling wine, such as the variety coming from the Champagne region of France, is a beverage that is at least partially famous for its carbon dioxide bubbles, a byproduct of the secondary fermentation process that occurs after bottling. A well-known theory, though hardly accepted universally, posits that the quality of a sparkling wine can be ascertained from the characteristics of its bubbles, such as bubble size distribution and rate of production. This talk describes a preliminary investigation to monitor the characteristics of sparkling wine bubbles using passive acoustic measurements, wherein bubble parameters are estimated from the power spectral density of the ambient bubble noise. Measurements made on a variety of sparkling wines will be presented.

1:15

2pPA2. Temperature and frequency dependent behavior of high intensity focused ultrasound (HIFU)-induced shear waves in a viscoelastic micellar fluid. E. G. Sunethra K. Dayavansha, Cecille Labuda, and Joel Mobley (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS 38677, sdayavan@go.olemiss.edu)

Wormlike micellar (WM) fluids, which flow when subjected to long term stresses, are mechanically viscoelastic over shorter durations. These fluids are birefringent under shear, allowing the study of shear wave propagation using both optical and acoustic modalities. In this work, the thermal and spectral behavior of ultrasonically generated shear waves in a WM fluid are studied. A fluid consisting of hexadecyltrimethylammonium bromide (CTAB) and sodium salicylate (NaSAL) combined in a 5:3 ratio is used in a 200 mM concentration. A high intensity focused ultrasound (HIFU) beam generates radiation pressure in the fluid and can induce shear waves of sufficient amplitude to be visualized optically when the beam is modulated. By pulsing the HIFU beam, a train of shear waves are generated which propagate laterally from the focal region. The temperature and frequency dependent behavior of the HIFU generated shear waves are correlated with the rheological and microstructural properties of the fluid.

1:30

2pPA3. Phase transition characterization based on acoustic reverberation time. Hossep Achdjian, Julien Bustillo, Laurianne Blanc, Andres Arciniegas, Nicole Doumit (GREMAN, Tours Univ., CNRS, INSA-CVL, Blois, France), and Marc Lethiecq (GREMAN, Tours Univ., CNRS, INSA-CVL, Bat E, 20 Ave., Monge, Parc de Grandmont, Tours 37000, France, marc.lethiecq@univ-tours.fr)

Non-destructive monitoring of a material's state during its physicochemical transformations is of interest for several industrial fields including food processing, such as milk-derived products, or cosmetics. The use of ultrasound to provide reliable information about physicochemical properties is becoming increasingly popular. Indeed, ultrasonic techniques have the main advantage of being rapid and non-invasive methods that allow parameters such as product composition, structure, and physical state to be obtained. Yet, classical techniques are limited to the characterization of the medium along the propagation path using the first wave packets. Here, we propose an alternative technique based on the study of the reverberated waves, classically used in room acoustics. In previous works, this method has shown its capability to characterize materials as well, with the advantage over classical techniques to address a medium in its whole structure. In this context, the determination of sol-gel phase transition time of Salol using this method is presented. Measurements are performed in an aluminum mold using five piezoelectric (PZT) patches, one being used as emitter and the others as receivers. The mean reverberation time over four receivers has been studied and its evolution is shown to lead to a good estimation of the phase transition time.

1:45

2pPA4. Rapid and simple measurement of hypersonic wave velocity by Brillouin scattering with induced phonons. Yoshiaki Shibagaki (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, duq0358@mail4.doshisha.ac.jp), Shinji Takayanagi (Nagoya Inst. of Technol., Nagoya, Japan), Masahiko Kawabe (Doshisha Univ., Kyotanabe, Japan), Takahiko Yanagitani, Masashi Suzuki (Waseda Univ., Tokyo, Japan), and Mami Matsukawa (Doshisha Univ., Kyotanabe, Japan)

Brillouin scattering is a non-contact method to measure wave velocities in the GHz range. One problem of the Brillouin scattering technique is weak light scattering from thermal phonons, which results in the long measurement time and necessity of a complex tandem Fabry-Perot interferometer. To overcome this problem, we have proposed techniques to make use of induced strong coherent phonons from a high frequency transducer. In this study, we have tried to induce strong longitudinal coherent phonons by a ScAlN film transducer (composition of the film : $Sc_{0.41}Al_{0.59}N$), which has a high electromechanical coupling coefficient. The transducer was fabricated on the quartz sample and composed of the ScAlN film grown by an RF magnetron sputtering and electrodes. The transducer was deposited on one side of the sample. Due to the induced phonons, the scattered light became much stronger than those of thermal phonons and could be observed by a simple confocal Fabry-Perot interferometer. The measured frequency shift of the Brillouin scattering peak was equal to the excitation frequency of the ScAlN transducer (883 MHz). This technique enables easy, rapid and simple measurement of wave velocity in the GHz range, which can be applied for the 2D velocity imaging of the sample.

2:00

2pPA5. Resonant ultrasound spectroscopy measurement of elastic properties of SnSe. Ashoka Karunarathne, Josh R. Gladden, Gautam Priyadarshan (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38677, atholang@go.olemiss.edu), Pai-Chun Wei, Yang-Yuan Chen (Inst. of Phys., Academia Sinica, Taipei, Taiwan), Sriparna Bhattacharya, and Apparao M. Rao (Dept. of Phys. and Astronomy, Clemson NanoMater. Inst., Clemson Univ., Clemson, SC)

Resonant Ultrasound Spectroscopy (RUS) is a precise experimental approach for investigating the elastic properties of solid materials at different temperatures and hydrostatic pressures. In RUS, the elastic stiffness tensor of crystalline solids with different crystal structures are determined from their vibrational resonance spectra. A study of the mechanical properties of thermoelectric materials can provide insights to their high efficiency and low thermal conductivity. SnSe is a thermoelectric material which exhibits a high efficiency and low thermal conductivity due to its anharmonicity and low symmetric crystal structure. SnSe exhibits a structural phase transition from *Pnma* to *Cnma* at $\sim$800 K, and the single layered orthorhombic crystal structure of SnSe results in nine independent elastic constants. In this study, RUS was used to determine the temperature dependent elastic properties of polycrystalline SnSe through its phase transition temperature. Use of RUS technique to better understand the temperature dependent elastic properties of low symmetry crystals will be discussed. The elastic constants of single crystal SnSe calculated by RUS at room temperature are in agreement of within $\sim$35% of theoretical reported values.

2:15

2pPA6. Focal zone characteristics of stepped Fresnel and axicon acoustic lenses. Robert L. Lirette (Phys., Univ. of MS, 2400 Anderson Rd., Apt. 4, Oxford, MS 38655, rlirette@go.olemiss.edu) and Joel Mobley (Phys., Univ. of MS, University, MS)

Flat Fresnel and axicon acoustic lenses were developed and characterized both numerically and experimentally. Flat lenses have a compact profile leading to less attenuation and phase distortion in the bulk lens material. Fresnel lenses approximate spherical focusing to a fixed point whereas the axicon focuses to a narrow focal line, producing a tighter lateral beam over a longer depth. Numerical models of the sound pressure field from both types of lenses were done using the angular spectrum method. The lenses were 3d printed with polylactic acid (PLA) and also lathe cut with transparent polystyrene. Pulse-echo measurements of the sound speed were done for both materials. Pressure field scans were conducted using a 1.2 MHz planar transducer in a hydrophone scanning tank. These scans demonstrate the focusing effect of both types of lenses and are in agreement with the numerical model.

2:30

2pPA7. Focus control in the radial direction using an ultrasound liquid crystal lens. Yuki Shimizu (Graduate School of Sci. and Eng., Graduate School of Doshisha Univ., Tataramiyakodani1-3, Kyotanabe, Kyoto 610-0321, Japan, duq0360@mail4.doshisha.ac.jp), Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), Yuki Harada (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), Akira Emoto, and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

Nematic liquid crystal is widely used in optical devices such as liquid crystal displays and controlled by electric fields through the liquid crystal layer. Our group has proposed a control technique of liquid crystal molecular orientation using ultrasound vibration without indium tin oxide electrodes [1]. In this study, using ultrasound and nematic liquid crystal, we realized a variable focus liquid crystal lens. The liquid crystal lens has no

2p TUE. PM

transparent electrodes and consists of a simple structure; a nematic liquid crystal layer with a thickness of 50 μm is formed between two circular grass plates. An annular PZT ultrasonic transducer was bonded on the glass plate. By exciting the transducer at the resonance frequencies (28 kHz), the flexural vibration modes were generated. The acoustic radiation force from the vibration changed the molecular orientation of the liquid crystals, which induced the distribution of the execution refractive indices. The liquid crystal layer then worked as a lens. The focal points could be controlled by the radiation force and moved in the radial direction. [1] S. Taniguchi, D. Koyama, Y. Shimizu, A. Emoto, K. Nakamura, and M. Matsukawa, *Appl. Phys. Lett.* **108**(2016), 101103.

2:45

2pPA8. Porous lead zirconate titanate ceramics with optimized acoustic properties for high-frequency ultrasonic transducer applications. Marc Lethiecq (GREMAN UMR 7347, Tours Univ., CNRS, INSA-CVL, Bat E, 20 Ave. Monge, Parc de Grandmont, Tours 37200, France, marc.lethiecq@univ-tours.fr), Danjela Kuscer (Electron. Ceramics, Jozef Stefan Inst., Ljubljana, Slovenia), Julien Bustillo (GREMAN UMR 7347, Tours Univ., CNRS, INSA-CVL, Blois, France), Andre-Pierre Abellard (Ctr. des Matériaux UMR7633 CNRS, Mines Paris Tech, Evry, France), Tina Bakaric (Electron. Ceramics, Jozef Stefan Inst., Ljubljana, Slovenia), and Franck Levassort (GREMAN UMR 7347, Tours Univ., CNRS, INSA-CVL, Tours, France)

Acoustic properties of porous PZT ceramics intended to be used as backing in high-frequency transducer applications are investigated using a novel method where an electroded piezoelectric thick film is deposited on the backing under test. Two backings with pore sizes 1.5 μm and 10 μm were obtained by sintering a mixture of ceramic powder and an organic template, their porosity was evaluated by scanning electron microscopy at 15%, leading to a density around 6.5 g/cm^3. The electroacoustic impulse responses of these devices were measured considering the backing as a propagation medium, the initial thickness of which was chosen small enough to allow back-wall echoes to be detected and large enough to be able to separate the signals in time domain. Then the thickness of the backing was reduced (from around 2 mm to less than 1 mm) and the measurements were repeated. Acoustic properties were then deduced: attenuation coefficients reaching 4 dB/mm/MHz and group velocities around 3400 m/s were obtained, leading to an acoustic impedance around 22 MRa. Such combination of high attenuation and moderate acoustical impedance make these materials an interesting solution for high-resolution ultrasonic imaging transducers.

TUESDAY AFTERNOON, 5 DECEMBER 2017 SALON A/B/C, 1:00 P.M. TO 4:30 P.M.

Session 2pSA

Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics: Acoustic Metamaterials II

Christina J. Naify, Cochair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr., Pasadena, CA 91109

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

1:00

2pSA1. Zone folding induced topological insulators in phononic crystals. Yuanchen Deng (North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695), minghui lu (Nanjing Univ., Nanjing, Jiangsu, China), and Yun Jing (North Carolina State Univ., Raleigh, NC, yjing2@ncsu.edu)

This study investigates a flow-free, pseudospin-based acoustic topological insulator. Zone folding, a strategy originated from photonic crystal, is used to form double Dirac cones in phononic crystal. The lattice symmetry of the phononic crystal is broken by tuning the size of the center "atom" of the unit cell in order to open the nontrivial topological gap. Robust sound one-way propagation is demonstrated both numerically and experimentally. This study provides a flexible approach for realizing acoustic topological insulators, which are promising for applications such as noise control and waveguide design.

1:15

2pSA2. Acoustic performance of phononic crystals for underwater coating applications. Gyani Shankar Sharma (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, Sydney, NSW 2052, Australia, gyanishankar.sharma@student.unsw.edu.au), Alex Skvortsov, Ian MacGillivray (Maritime Div., Defence Sci. and Technol. Group, Melbourne, VIC, Australia), and Nicole Kessissoglou (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, Sydney, NSW, Australia)

Acoustic coatings on maritime vehicles can significantly reduce the transmission of machinery noise in the ambient marine environment as well as absorb external acoustic waves. In this work, the performance of two types of phononic crystals with steel backing is investigated for acoustic coating applications. The first type of phononic crystal comprises periodic voids embedded in a soft elastic medium. The second type of phononic crystal comprises hard scatterers arranged periodically in an elastic medium.

The voids exhibit monopole resonance, leading to low sound transmission through the coating in a broad frequency range. In contrast, the hard scatterers exhibit dipole resonance which results in low sound reflection. The ratio of bulk to shear moduli of the elastic medium governs the monopole resonance, whereas the dipole resonance is governed by the ratio of the density of the scatterers to the density of the host medium. The effect of steel backing on the transmission side of the elastic medium results in high values of sound absorption attributed to Fabry-Perot resonance. The advantages and limitations of the two types of acoustic coatings are discussed.

1:30

2pSA3. Optimizing sound transmission through 3D pentamode materials in water. Xiaoshi Su and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Pentamode materials (PMs) are engineered isotropic or anisotropic elastic structures that mimic the acoustic behavior of ideal fluids. Recently, many applications such as negative refraction lens, focal lens, and a cloak have been designed and demonstrated using 2D PMs. We consider metallic lattice structures in which only one wave mode propagates over a wide frequency band. Unlike the 2D PM which naturally has a boundary, the three dimensional PM needs to be sealed to isolate the interior structure from water. However, the material used for encapsulation alters the effective surface impedance of the material. In this talk, we present 3D PMs with acoustic properties matched to water and show that the transmission is sensitive to the thickness of the boundary plate. An analytic model is established to provide some insights into the boundary transmission. Iterations were performed in FEM simulations to achieve optimal transmission within the frequency range of interest (4 kHz–10 kHz). The model PM with partial boundary is made using 3D metal printing, and the remaining boundary is added post fabrication. Preliminary experimental results will be discussed. [Work supported by ONR/MURI.]

1:45

2pSA4. Numerical investigation of metafluid behavior in pre-strained negative stiffness honeycombs. Benjamin M. Goldsberry and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu)

Negative stiffness honeycombs (NSH) are doubly periodic arrays of curved beams that have recently been shown to exhibit a reversible pseudoplastic large deformation response and are therefore interesting candidates for impact isolation applications. The authors previously used finite element analysis to show that NSH exhibit three types of propagating modes described by (i) longitudinal, (ii) transverse, and (iii) rotational displacements with respect to the propagation direction. The phase and group velocities of each mode can be significantly altered by applying a uniaxial pre-strain imposed on the top and bottom boundaries of the NSH. The present work will show that the transverse and rotational mode phase speeds are significantly more sensitive to strain than the longitudinal modes for small but finite pre-strain levels. Consequently, for some pre-strain levels there exists a large frequency band where only longitudinal waves propagate. The pre-strained NSH therefore behaves as a pentamode metamaterial, or metafluid, and may represent a new type of structure to realize devices designed using transformation acoustics. The effect of the geometry and pre-strain on the existence and frequency range of unimodal propagation in pre-strained NSH will be investigated. [Work supported by NSF.]

2:00

2pSA5. Enhancement of low-frequency sound emission by metamaterial enclosures. Likun Zhang (Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu), Jiajun Zhao (King Abdullah Univ. of Sci. and Technol. (KAUST), Austin, Texas), Ying Wu (King Abdullah Univ. of Sci. and Technol. (KAUST), Jeddah, Saudi Arabia), and Maryam Landi (Dept. of Phys. and Astronomy, Univ. of MS, Oxford, MS)

Emission of low-frequency sound by a source is limited by the size of the source relative to sound wavelength. We propose to place the source in a subwavelength enclosure of anisotropic metamaterials which enhances the emission around Mie resonant frequencies of that enclosure [J. Zhao, L. Zhao, and Y. Wu, *J. Acoust. Soc. Am.* **142**(1), EL24–29, July 2017]. The enclosure has a relatively small sound speed along the radiation direction, enabling the resonances to occur at low frequencies. Our numerical simulations provide evidence of the proposed low-frequency enhancement for both monopole and multipole sources. The common Mie resonant frequencies increase with the order of the multipoles. Here, we introduce an extreme anisotropy in the azimuth of the enclosure to degenerate the resonant frequencies of higher-order multipoles down to the same frequencies as the lower-order multipoles. The degeneracy associated with the anisotropy of the enclosure is theoretically analyzed. The results guide the experimental realization of the enclosure for low-frequency enhancement.

2:15

2pSA6. Experimental realization of enhancing low-frequency sound emission by metamaterial enclosure. Maryam Landi, Likun Zhang (Univ. of Mississippi/National Ctr. for Physical Acoust. (NCPA), 145 Hill Dr., University, MS 38677, mlandi@go.olemiss.edu), Jiajun Zhao (Div. of Comput., Elec. and Mathematical Sci. and Eng., King Abdullah Univ. of Sci. and Technol. (KAUST), Austin, Texas), and Ying Wu (Div. of Comput., Elec. and Mathematical Sci. and Eng., King Abdullah Univ. of Sci. and Technol. (KAUST), Jeddah, Saudi Arabia)

A solution to the low radiation rate of a sound source has been proposed by using metamaterial enclosures of degenerated Mie resonances [J. Zhao, L. Zhao, and Y. Wu, *J. Acoust. Soc. Am.* **142**(1), EL24–29, July 2017]. This talk presents our experimental realization of this enhancement for a monopole source. The enclosure with a low sound speed is created by a coiling-space structure fabricated by 3D printing. A small balanced-armature speaker with a 1 mm speaker mouth is placed at the enclosure center to radiate impulsive sound signals. The diameter of the structure is only one seventh of the wavelength of the sound at the maximum enhancement, corresponding to the lowest resonance. The enhanced sound pressure is more than two orders of magnitude larger than the sound pressure without the enclosure. The measurements of sound pressure at different angles in the space show the omnidirectionality of the enhancement. This enhancement is demonstrated as a result of enhanced density of states, i.e., the increased number of modes per unit frequency range and per unit volume, which we relate to Fermi's Golden rule and Purcell effect in quantum systems and find the analog in acoustics.

2:30–2:45 Break

2:45

2pSA7. Compressibility-near-zero acoustic supercoupling. Matthew S. Byrne (Elec. and Comput. Eng., Univ. of Texas at Austin, 1616 Guadalupe St., Austin, TX 78751, mbyrne@utexas.edu), Hussein Esfahlani (Ecole Polytechnique Fédérale de Lausanne, Lausanne, Switzerland), Matthew J. McDermott (Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Andrea Alu (Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX)

Epsilon-near-zero supercoupling is a widely-researched topic in electromagnetics. This phenomenon takes advantage of media with near-zero dielectric permittivity to build waveguide coupling channels which can, in principle, support unitary transmission and complete phase uniformity, independent of the length and height of the coupling channel. In this effort, we present the possibility that an analogous, extraordinary coupled-mode phenomenon exists in acoustics, which we call compressibility-near-zero (CNZ) supercoupling. Prior works have required the presence of periodic membrane resonances in order to observe near-zero index coupling. We have shown, for the first time, theoretically and experimentally, that the phenomenon can be observed without the use of membranes. We envision this development as a first step towards experimentally realizing spatiotemporally-modulated acoustic non-reciprocal devices which take advantage of these physical principles.

3:00

2pSA8. Acoustic cloak by inverse design and gradient based optimization. Feruza Amirkulova (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza.amirkulova@wne.edu) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

We present a gradient based minimization of the total scattering cross section (TSCS) from a set of cylindrical obstacles by incrementally repositioning them so that they eventually act as an effective cloaking device. The idea differs from earlier inverse designs that use topology optimization tools and generic algorithms. We use the optical theorem to define the position dependent TSCS in terms of the forward scattering amplitude for an incident plane wave. The gradient-based optimization algorithm reduces the TSCS by evaluating its derivative with respect to the cylinder positions and then perturbatively optimizing the position of each cylinder in the cloaking device while taking into account acoustic the multiple scattering between the cylinders. The method is illustrated by examples of sets of hard cylinders of uniform size.

3:15

2pSA9. Unidirectional sound manipulation with acoustic metamaterials. Jie Zhu (Hong Kong Polytechnic Univ., Kowloon 00000, Hong Kong, jiezhu@polyu.edu.hk)

Abstract: Unidirectional sound propagation introduces strong direction-dependent responses. Such effect has drawn considerable attentions in acoustics. In this presentation, we introduce some of the interesting unidirectional sound manipulation phenomenon that is achieved by acoustic metamaterials. The experimentally observed phenomenon is associated with the enhanced wave-structure interactions. We can predict and tune the operating frequency by further trimming the material's geometry. Our demonstration provides a new degree of freedom to the realization of unique wave dynamics for applications like noise control, acoustic sensing, and imaging. [This work was supported by the Early Career Scheme (ECS) of Hong Kong RGC (Grant No. 25208115).]

3:30

2pSA10. Double-decorated membrane structure: Initial efforts towards real-time phase modulation. Reza Ghaffarivardavagh (Mech. Eng., Boston Univ., 8 saint Mary St., Photonic Bldg., Rm. 832, Boston, MA 02215, ghaffari@bu.edu), Stephan Anderson (Radiology, Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., Boston, MA)

One of the fundamental challenges in the practical implementation of acoustic metamaterials science is the issue of tunability. To date, proposed acoustic metamaterial structures and devices suffer from narrow-band working frequency are typically passive structures. For example, with regard to phase modulation using metasurfaces, the tunability and capability of real-time phase modulation is critical in numerous applications ranging from biomedical ultrasound to acoustic communication. The study presented herein represents an initial step toward realizing acoustic metamaterials with real-time tunability intended for phase modulation applications. Herein, the double-decorated membrane structure (DDM) will be introduced, in which fundamental resonance of two membranes have been tailored to generate constructive interference, enabling phase modulation across a large range, while the amplitude variation has been mitigated. In this study, analytical and numerical investigations have been conducted to explore and optimize the design parameters. The proposed structure can be also used for the aim of low frequency sound energy harvesting in which the DDM structure will broaden the working frequency band and, consequently, enhance the energy conversion efficiency.

3:45

2pSA11. A new approach to generate local resonator for the application of acoustic/elastic metamaterials. Sheng Sang (System Eng., Univ. of Arkansas at Little Rock, 1701 Westpark Dr., Apt. #110, Little Rock, AR 72204, sxsang@ualr.edu)

In this paper, we proposed a type of locally resonator which works by introducing more than one vibration modes, and thus can provide a new approach to formulate acoustic or elastic metamaterial. Such new concept of resonator is fundamentally different from traditional translational and rotational resonators. The potential application and design of acoustic metamaterial based on this kind of structures are also studied both analytically and numerically. The acoustic metamaterial filter we developed has proved to have unique property that only allow the waves with specific frequency to pass while other waves will be attenuated. We also proposed a kind of plane wave lens which can transfer plane wave with uniformly distributed amplitude profiles into pure plane wave. With this novel resonator, the research and application regarding it are highly anticipated in the near future.

4:00

2pSA12. Active vibration control by a tunable piezoelectric shunt metamaterial. Aaron J. Stearns (Mech. Eng., Applied Res. Lab., Penn State Univ., 225 Sci. Park Rd., State College, PA 16803-2213, ajs6037@psu.edu) and Benjamin Beck (Structural Acoust., Appl. Res. Lab., Penn State Univ. State College, PA)

Acoustic metamaterials are composite materials exhibiting effective properties and acoustic behavior not found in traditional materials. Primarily through periodic subwavelength resonant inclusions, acoustic metamaterials can enable steering, cloaking, lensing, and frequency band control of acoustic waves. However, a common drawback of acoustic metamaterials is that effectiveness is limited to narrow frequency bands. Thus, investigation of practical active and adaptable acoustic metamaterials is valuable in achieving wider operation frequency bands. Here, a metamaterial consisting of active tunable piezoelectric shunts is investigated numerically and experimentally from the unit cell level. A physical model of the unit cell is developed using the finite element method. From the finite element model, the wave finite element method is applied to compute the dispersion and forced response of the periodic structure. It is demonstrated that the shunts introduce an additional degree of freedom by which adaptable bending wave attenuation can be accomplished. Since the periodic shunts are only effective at certain frequency bands, a known optimization method is implemented to tune the shunts. Additionally, a new optimization scheme is compared to the existing scheme found in literature.

4:15

2pSA13. Digital design of cellular solids for noise and vibration mitigation. Mark J. Cops, James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, mcops@bu.edu), and Elizabeth A. Magliula (NAVSEA, Newport, RI)

Cellular solids are of large interest in structural vibration due to their high strength-to-weight ratio and ability for high energy absorption. This presentation describes concepts for digitally designing cellular solids and demonstrates the ability of this design method for tuning effective material properties for noise and vibration mitigation applications. The designs can be created by defining a complete topology mathematically or by specifying section cut-outs from a solid host material. The digital designs are then analyzed with finite element software to determine effective material properties. This approach is advantageous because it allows for an automated computer procedure for designing and characterizing materials. In addition, resulting designs can be fabricated through either 3D printing processes or CNC milling. In this presentation, the relationship between geometrical/physical properties and effective static material properties (such as Young's modulus and Poisson ratio) of cellular solids will be discussed for the digital designs, with reference to existing theory.

Session 2pSC

Speech Communication: Speech Production (Poster Session)

Meghan Clayards, Chair
McGill University, 1085 Ave., Dr. Penfield, Montreal, QC H3A 1A7, Canada

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

2pSC1. Tracking larynx movement in real-time MRI data. Miran Oh (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu), Asterios Toutios (Elec. Eng., Univ. of Southern California, Los Angeles, CA), Dani Byrd, Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Ejective and implosive consonants involve rapid raising or lowering, respectively, of the larynx coordinated with an oral constriction formation and release. Techniques such as contour tracking and region-of-interest (ROI) direct image analysis are becoming established for examining supralaryngeal constriction formation in real-time MRI data, but quantification of larynx movement from dynamic vocal-tract magnetic resonance imaging has not been tackled. We will evaluate methods of indexing the movement of the larynx in fast real-time MRI data collected in the midsagittal plane. We analyze data from USC's real-time MRI IPA database produced by several phoneticians and data from speakers of the Hausa language producing both ejectives and implosives. Kinematic profiles of the laryngeal movements are obtained by outlining the arytenoid structure and applying principal component analysis on the vocal-tract outlines. In addition, vertical movement of the larynx is indexed by calculating the time-varying pixel intensity centroid (i.e., intensity-weighted average spatial position) in a rectangular ROI defined for larynx movement. Once direct measurement of the larynx actions is validated for this type of imaging data, further work will focus on the temporal coordination of the laryngeal raising/lowering gestures and the supralaryngeal constriction gestures that occur in glottalic airstream consonants. [Work supported by NIH.]

2pSC2. Toward a dynamic theory of vowel production. Fang Hu (Inst. of Linguist., Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn)

Vowels are traditionally classified into monophthongs, diphthongs, triphthongs, and in some rare cases, tetraphthongs. However, there is a long debate on the complexity of vowels in the literature. Some phoneticians view diphthongs as a single vowel with phonetically complex nucleus, while others treat diphthongs as a sequence of two vowel elements. This paper reports recent developments in vowel researches in Chinese dialects, and argues for an integral account for vowel production. First, fine-gained phonetic details from Mandarin, Wu, Jin, Hui, Min, and Hakka reveal that falling diphthongs and rising diphthongs differ in temporal organization, spectral property, and spectral dynamics. Second, there is no dichotomy, but a continuum between monophthongs and diphthongs. Comparisons between four Hui dialects reveal how the process of diphthongization develops from a more monophthong-like stage to a more diphthong-like stage. In conclusion, the production data from Chinese dialects support an integral account for vowel production. Monophthongs are composed of a static spectral target, diphthongized vowels and falling diphthongs are composed of a dynamic spectral target, and rising diphthongs are sequences of two spectral targets.

2pSC3. Using machine learning to identify articulatory gestures in time course data. Will Styler, Jelena Krivokapic (Dept. of Linguist, Univ. of Michigan, 611 Tappan St., 440 Lorch Hall, Ann Arbor, MI 48109, wstyler@umich.edu), Ben Parrell (Linguist and Cognit. Sci., Univ. of Delaware, Newark, DE), and Jiseung Kim (Dept. of Linguist, Univ. of Michigan, Ann Arbor, MI)

One difficulty in working with articulatory data is objectively identifying phonological gestures, that is, distinguishing targeted gestural movement from general variability. Although human annotators are generally used, an automated approach to identifying meaningful patterns offers advantages in speed, consistency, and objective characterization of gestures (cf. Shaw and Kawahara 2017). This study examines Electromagnetic Articulography (EMA) data from seven American English speakers, aiming to identify and characterize pause postures (specific vocal tract configurations at prosodic boundaries; Katsika *et al.* 2014). Supervised machine learning using kernelized Support Vector Machine Classifiers (SVMs) took as training data 852 trajectories from three speakers analyzed to date, containing 104 pause postures identified by a human annotator, and classified the between-words lip aperture (LA) trajectory to identify tokens containing the pause posture, while also providing token-by-token gesture probability. Features of the curvature were extracted using Principal Component Analysis (PCA) and Discrete Cosine Transform (DCT) of both the actual LA trajectory and of the deviation from a direct word-to-word interpolation. The SVM achieves 94.0% classification accuracy in cross-validation tests, with Cohen's Kappa showing machine-to-annotator agreement of 0.978. These methods of machine-learning-based curve classification are potentially useful and applicable to any time-course articulatory data. [Work supported by NIH and NSF.]

2pSC4. Distributional factors in Telugu sibilant production. Charles Redmon, Allard Jongman, and Jie Zhang (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., Rm. 427, Lawrence, KS 66046, redmon@ku.edu)

Telugu is one of a small set of languages described as exhibiting three or more place contrasts among sibilant fricatives (less than 4% of languages in UPSID; Maddieson and Precoda, 1990). In a pilot study of alveolar, palatal, and retroflex sibilant productions in VCV sequences, Telugu speakers were found to show consistent inter-speaker variation in the production of the palatal sibilant, with half showing evidence (in production and perception) of a complete merger with the alveolar, and the other half showing evidence of a merger with the retroflex, and no clear demographic differences underlying the delineation of the two groups. Following up on this result, we present acoustic data from native speakers of Telugu recorded in Hyderabad producing words with the three sibilants in multiple vowel contexts,

positions (CV, VC), and lexical neighborhood structures (with vs. without sibilant competitors in a second order neighborhood), focusing in the acoustics in particular on formant transition cues to palatal and retroflex places. This data are presented both to clarify the present state of the Telugu sibilant system and to explore what effect the lexical distribution of these sounds has on the acoustic feature mapping of the contrast.

2pSC5. Uniformity of inherent vowel duration across speakers of American English. Colin Wilson (Johns Hopkins, Krieger 247, 3400 North Charles St., Baltimore, MD 21218, colin.wilson@jhu.edu) and Eleanor Chodroff (Northwestern Univ., Evanston, IL)

Vowel duration is determined by a number of factors in American English (e.g., Klatt, 1976), including the tense vs. lax distinction (e.g., /i/-/ɪ/, /u/-/ʊ/) and the relation between duration and vowel height (e.g., Lehiste, 1970). A large body of research has identified segment-internal factors in data averaged over many speakers (e.g., Crystal and House, 1988; Hillenbrand *et al.*, 1995), but few studies have investigated the inherent vowel duration patterns of individuals (cf. House, 1961). Stressed vowel productions (>2 million tokens) were identified by forced alignment in connected speech recordings of 390 speakers (209 female) from the Mixer 6 corpus (Chodroff *et al.*, 2016). For each speaker, mean durations of ten stressed vowels (/i ɪ e ɛ æ a ʌ o ʊ u/) were calculated after outlier exclusion. The resulting duration patterns were strongly correlated across pairs of speakers (Pearson *r*: mean = 0.936, 95% CI [0.935, 0.936], range [0.559, 0.999]), and PCA identified a single component, plausibly indexing global speaking rate, that accounted for more than 82% of the variance. These results establish that inherent vowel duration is highly uniform across speakers, a type of structured phonetic variation that has implications for models of perceptual adaptation.

2pSC6. Phonological neighborhood density and intra-speaker variation in vowel production. Benjamin Munson, Tait Anderson, and Jayanthi Sasisekaran (Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Words' phonetic characteristics are affected by numerous lexical factors. Wright (2004) showed that words' phonological neighborhood density (PND) affects the production of vowels. Vowels in words with high PND are hyperarticulated relative to the same vowels in low-PND words. Wright interpreted this as evidence that speech production accommodates listeners' presumed needs: high-PND words are hyper-articulated because they are generally harder to perceive than low-PND words (Vitevitch & Luce, 1998). This effect has been the subject of numerous recent studies, including ones by Munson and Solomon (2004) and Munson (2013) that replicated the effect, and ones by Buz and Jaeger (2016) and Gahl, Yao, and Johnson (2012) that did not. One possible explanation for Wright's original effect is that the ease with which low-density words are perceived allows speakers more freedom to produce the sounds in them variably, including under-articulating them. If this is true, then we would predict that the acoustic characteristics of low-PND words would show more trial-to-trial variability than those of high-PND words. To examine this, we recorded 19 speakers saying 10 repetitions each of high- and low-PND CVC words. The F1 and F2 frequencies of the vowels were measured. For each token, the distance from the center of each speaker's F1/F2 space was computed. The coefficient of variation for this measure was higher (i.e., more variable) for low-PND words than for high-PND words, $t[18]=-2.80$, $p=0.011$.

2pSC7. Acoustic features of spontaneous and read conversational and clear speech produced in simulated acoustic environments. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), and Jessica J. Staples (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Speech production can differ depending on how speech is elicited (e.g. spontaneous speech, read text, speaking style instructions, the speaking environment). In most studies in which different speaking styles have been elicited via instruction (e.g., clear speech) or via the speaking environment (e.g., Lombard speech), the talkers have read printed materials. There is evidence that the acoustic features of clear speech elicited by reading are similar to those observed in semi-spontaneous interaction between two interlocutors, but that clear speech changes are of a greater magnitude in the read speech than the semi-spontaneous speech [V. Hazan and R. Baker, *J. Acoust. Soc. Am.* **130**(4), 2139–2152, 2011]. The present study examines the effects of two speaking style instructions (conversational and clear) and four simulated listening environments (quiet, 55 dB SPL of white noise, 63 dB SPL of white noise, and a reverberant environment) presented via earphones for three types of speech materials: read sentences, read passages, and spontaneously produced picture descriptions. Acoustic features relevant to clear speech and Lombard speech will be compared among the three material types.

2pSC8. Vocalic processes in Southern Ute. Viktor Kharlamov (Florida Atlantic Univ., 777 Glades Rd., CU-97, Ste. 280, Boca Raton, FL 33431, vkharlamov@fau.edu) and Stacey Oberly (Southern Ute Indian Montessori Acad., Ignacio, CO)

Most of the Native American languages are severely endangered and lack detailed phonetic descriptions. Our research aims to document the sound system of Southern Ute, a Numic language of the Ute-Aztecan family spoken by approximately 40 elders in southwestern Colorado, for which only basic impressionistic descriptions are available in the published grammar of the language (Givon, 1980). In this acoustic study, we focus on Southern Ute vowels. We analyze over 6,000 vowel tokens produced by 8 fluent speakers and present our findings on a variety of vocalic processes in Southern Ute, including positional allophony, vowel harmony, as well as phonetic aspects of lexical stress and durational differences related to the phonological length distinction.

2pSC9. Determining the place of articulation of Arabic voiceless dorsal fricatives. Sherman D. Charles, Abdullah Alfaifi, Noor Abo-Mokh, Sarah Robinson, and Kenneth de Jong (Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Previous, mostly impressionistic, examinations of Arabic varieties provide varied descriptions of the dorsal fricative /χ/ place of articulation, ranging from velar to uvular. While some descriptions attribute this variation to dialectal differences, descriptions of the same dialect also vary. 3D ultrasound data were collected from six native speakers of different dialects of Arabic, one each from Syria, Palestine, Faifi (in Saudi Arabia), Egypt, Algeria, and Morocco. Apart from the dorsal fricatives, the corpus included productions of palatal, pharyngeal, and contrasting velar and uvular stops to provide comparative standards for various points of articulation. The Syrian, speaker showed very similar articulations for /χ/ with uvular stops, while the other speakers variably showed a more anterior articulation between the velar and uvular stops. The most anterior articulations were apparent in the Moroccan and Algerian speakers, giving some suggestion of a dialectal difference; however, point of articulation of the dorsal fricative was variable for most of the speakers and not obviously restricted by dialect. The effect of these articulatory observations on the acoustic noise spectra are ongoing.

2pSC10. Vowel-to-vowel coarticulation in Spanish non-words: Effects of stress and consonantal context. Jenna Conklin and Olga Dmitrieva (Linguist, Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu)

In spoken language, gestural overlap in speech production regularly leads to coarticulation between neighboring segments, resulting in assimilation measurable by changes in the acoustic parameters. Vowels in adjacent syllables can coarticulate in a phenomenon called vowel-to-vowel coarticulation, which is subject to variation based on environmental factors such as surrounding consonantal context and the placement of stress. This study investigates vowel-to-vowel coarticulation in Spanish in order to better understand the effect of stress and consonantal context on coarticulation. Formant analysis of vowels produced by 20 native speakers of Spanish was used to determine the presence and direction of coarticulation in trisyllabic nonce words with varying stress (/CVCVCV/ words, vowels /e, o/ as targets and /e, i, o, u/ as triggers, /k/ and /p/ contexts). Results showed that both anticipatory and carryover vowel-to-vowel coarticulation was present at vowel

edges in all contexts, but only carryover coarticulation in backness extended to vowel midpoints. Stress placement mediated the coarticulatory effect with select target and trigger combinations in carryover coarticulation, while consonantal context played a greater role in anticipatory coarticulation. When stress played a role, unstressed target vowels were more susceptible to coarticulation, while stressed targets resisted coarticulatory effects.

2pSC11. The phonetic effects of onset complexity on the English syllable. Anna Mai (Linguist, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, acmai@ucsd.edu)

In English, onset segments of the syllable do not participate in the categorical criteria which determine phonological syllable weight, yet they probabilistically impact stress assignment in novel words and influence the composition of weight-based metric verse (Kelly 2004, Ryan 2014). Together these observations suggest a phonetic motivation for the role that onsets play in English's phonological syllable weight system, investigated here. Using 20 rhyming sets of single syllable words minimally differing in onset material (i.e., *rap, trap, strap*), this production study replicates results from the literature showing that the addition of segmental material to the onset accompanies a decrease in vowel duration (Gordon 2005, Ryan 2014). Furthermore, results show that the pitch and amplitude maxima of the syllable occur earlier within syllables containing more onset segments. Pitch and amplitude maxima have previously been implicated in perceptually based accounts of syllable weight (Goedemans 1998), and since stimuli were controlled for stress and categorical weight, these results suggest that the phonetic effects observed reflect general acoustic properties of onset complexity in the language. For these reasons, findings reported here suggest that perceptual acoustic effects intrinsic to onset complexity are exploited by the weight system of English despite exclusion from its categorical criteria.

2pSC12. Acoustics of the tense-lax stop contrast in Semarang Javanese. Scott Seyfarth (Linguist, NYU, New York, NY), Jozina Vander Klok (Linguist and Scandinavian Studies, Univ. of Oslo, Oslo, Norway), and Marc Garellek (Linguist, UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu)

Javanese has a contrast between tense and lax stops. While both tense and lax stops are voiceless and unaspirated, the contrast at least in word-initial position is realized through acoustic differences in the following vowel, including lower f0, breathier voice quality, and higher F1 for the lax stops relative to their tense counterparts. However, previous reports have indicated substantial cross-speaker variation, and in some cases involve differing characterizations of the acoustic contrast, possibly due to small sample sizes. Moreover, it is still unclear whether (and how) this contrast is maintained in word-final position. In this study, we investigate the tense-lax contrast based on audio recordings of 27 speakers of Central Javanese from Semarang, Indonesia who each read 30 or more items with a word-initial or word-final stop in a carrier phrase. Stops and their adjacent vowels were hand-annotated, and measurements were taken including voice onset and offset time, stop closure and release duration, f0, formant frequencies, and spectral tilt and noise levels. Discussion will focus on how the tense-lax contrast is realized in both word-initial and word-final positions, as well as how different acoustic correlates co-vary within and across speakers.

2pSC13. An electromagnetic articulography study of stop-/s/ and /s/-stop clusters in Greek. Evdoxia Doli, Patrick Reidy, and William F. Katz (Callier Ctr. for Commun. Disord., The Univ. of Texas at Dallas, 1966 Inwood Rd, Dallas, TX 75235, exd160330@utdallas.edu)

Previous kinematic studies on consonant clusters suggest that many factors affect their articulatory timing, including speech rate, frequency of occurrence, and prosody. Other possible factors include the direction of articulatory movement (front-to-back, back-to-front), whether independent articulators (e.g., tongue and lips) or single articulators (e.g., tongue tip and tongue body) are involved, as well as the relative sonority of the consonantal segments. In this study, five talkers of Modern Greek produced the clusters /sp/, /ps/, /sk/, and /ks/ in the carrier phrase "Ipa_____pali" (I said______again). These clusters varied in articulatory direction, articulatory independence, and sonority patterns. We used four methods to determine the degree of articulatory overlap. Each method yielded durations (ms) between gestural landmarks, which were used to compute consonant overlap. The methods differed in the ways gestures were interpreted from the velocity signal (e.g., using extrema or 20% threshold) and how consonant overlap was computed. Preliminary results indicate that articulatory direction and relative sonority exerted the strongest effects on speakers' productions, while articulator dependence/independence played no apparent role.

2pSC14. A case report of nasal fricatives with grimacing: Evidence for spectral marking of the /s/-/ʃ/ contrast. 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) and David J. Zajac (Dental Ecology, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

A three-year old girl with repaired bilateral cleft lip and palate was observed to produce anterior nasal fricatives (ANFs) with grimacing during video administration of GFTA-3. ANFs are learned maladaptive articulations used to replace oral fricatives. They are produced by occluding the oral cavity and forcing all airflow through an open velopharyngeal (VP) port that results in turbulent noise at the anterior liminal valve of the nose. Although the girl had normal hearing at the time of testing, she had a history of conductive hearing loss and ventilation tubes. Pressure-flow testing confirmed VP dysfunction and a small nasal area during breathing. The girl appeared to grimace more severely during production of targeted /ʃ/ as compared to /s/ sounds. Greater grimacing on /ʃ/ was confirmed by a forced-choice task with three raters who independently viewed randomized video segments of /s/-/ʃ/ words. Spectral moment analysis revealed higher spectral mean, more negative skewness, and higher kurtosis for ANFs substituted for /s/ as compared to /ʃ/ sounds. We conclude that the girl used a nasal grimace as an articulatory gesture—perhaps learned during speech therapy—to spectrally mark the /s/-/ʃ/ contrast by modulating the length and cross-sectional area of the anterior nasal valve. [Research reported in this publication was supported by the National Institute of Dental & Craniofacial Research of the National Institutes of Health under Award Number 1R01DE022566-01A1.]

2pSC15. Indexing tongue profile narrowing for English lateral consonants using 3D volumetric MR imaging. Mairym Llorens, Dani Byrd (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu), Nancy Vazquez (Linguist, California State Univ., Long Beach, CA), Louis Goldstein, Tanner Sorensen (Linguist, Univ. of Southern California, Los Angeles, CA), Asterios Toutios, and Shrikanth S. Narayanan (Eng., Univ. of Southern California, Los Angeles, CA)

Production of lateral consonants in many languages involves separate but coordinated tongue tip and tongue rear actions—raising of the tongue tip and retraction of the tongue body. Given these gestures and the presence of lateral airflow, it has been speculated that horizontal (i.e., side-to-side) narrowing of the tongue may occur during production of these laterals, either as a passive result of the anterior-posterior lingual stretching or as an actively controlled movement. This study uses 3D volumetric MR scans of speakers producing a variety of sustained English sounds to examine horizontal tongue width as a function of consonant laterality/centrality and as a function of vowel height in front vowel contexts. Multiplanar reconstruction of volumetric data for each token and subject permitted imaging of the static postures on the axial plane. We determine a protocol for identifying a specific oblique axial slice in terms of mid-sagittal anatomical landmarks that allows for a stable and informative index of horizontal tongue width. This index demonstrates utility for identifying and quantifying tongue profile narrowing specific to English lateral consonants. [Work supported by NIH.]

2pSC16. Quantifying labial, palatal, and pharyngeal contributions to third formant lowering in American English /ɹ/. Sarah Harper, Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, skharper@usc.edu), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Although the acoustic profile of /ɹ/ is largely stable across speakers and contexts in rhotic dialects of American English (e.g., Hagiwara, 1995), significant variability is observed in the articulatory implementation of the palatal, pharyngeal, and labial constriction gestures involved in its production (Delattre & Freeman, 1968; Alwan *et al*., 1999; Tiede *et al*., 2004). In order to quantify each gesture's relative contribution to F3 lowering, we expand upon previous work on articulatory-acoustic relations in American English /ɹ/ by examining the acoustic effect of articulatory variation separately for each supralaryngeal constriction gesture. Real-time MRI data from four speakers in the USC-TIMIT corpus (Narayanan *et al*., 2014) were analyzed to determine the location, length, and aperture for each constriction gesture in 668 tokens of /ɹ/. Acoustic data were taken from simultaneous audio recordings of each speaker's real-time MRI capture, with F1-F4 values extracted at the time of maximal constriction for each gesture. Our results suggest that each gesture's acoustic contribution to F3 depends on between- and within-speaker variation in the articulation of /ɹ/, as variation in the length and location of the palatal and pharyngeal gestures is associated with differences in their relative effect size on F3 lowering. [Work supported by NIH.]

2pSC17. On simultaneous electromagnetic articulography and electroglottography data acquisition. Sarah Harper, Sungbok Lee, Dani Byrd, and Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, skharper@usc.edu)

Simultaneous measurements of speech articulator movement and laryngeal activity are desirable for obtaining a picture of the coarticulatory behavior between oral articulators and laryngeal vocal fold behavior. However, since electroglottography (EGG) use the application of a weak current across the larynx, which is in close proximity to the oral vocal tract, there is a possibility that its use may interfere with the sensor voltage fluctuation measurements used in electromagnetic articulography (EMA). In order to investigate any hypothetical interference effect, data was collected using Northern Digital Instruments' Wave articulograph with Glottal Enterprises's EG2-PCX electroglottograph. Datasets were collected from one male and one female speaker—both with simultaneous EMA and EGG and, for comparison, with EMA alone. Means and standard deviations of inter-sensor distances for static (i.e., reference) and moving (e.g., tongue tip, lip, and jaw) EMA sensor trajectories were compared across the two collection conditions for possible interference effects. Preliminary results find no discernable differences in inter-sensor distances for either static or moving EMA sensors between the with- and without-EGG conditions. If this finding is maintained for additional speakers, NDI EMA data and GE EGG signals may be collected simultaneously without adverse effects on the measurement. [Work supported by NIH.]

2pSC18. The distribution of the retroflex lateral allophone of the liquid phoneme in Korean. Nari Rhee (Linguist, Univ. of Pennsylvania, 2930 Chestnut St., APT2405B, Philadelphia, PA 19104, n.rhee94@gmail.com)

The liquid phoneme in Korean is known to have an optional retroflex allophone in syllable-final positions (Lee 1999). However, there lacks literature evaluating when and how this retroflex allophone is realized. To define the distribution of the retroflex allophone of the liquid phoneme in Korean, this study uses a read speech corpus of Seoul Korean to track the rhotacization of the liquid phoneme in coda positions. Rhotacization of the liquid is detected and measured by the lowering of the third formant. Preliminary results show that the liquid phoneme in coda position is realized as retroflex when preceded by a back vowel, and when followed by another liquid phoneme (gemination). The results suggest that the liquid phoneme is not simply optionally rhotacized in coda positions; instead, the retroflex allophone is realized in contexts defined by the preceding vowel and the following consonant. Future research remains to further examine the articulatory and perceptual aspects of the rhotacization.

2pSC19. Articulatory and acoustic correlates of tongue root contrasts in Gua. Samantha Myers, Kelly Berkson, and Kenneth de Jong (Dept. of Linguistics, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine 844, Bloomington, IN 47405, kberkson@indiana.edu)

Vowel systems in West African languages are often noted for using the position of the tongue root (TR) to contrast vowels throughout the vowel space. E.g. X-ray studies of Igbo show that pairs of vowels such as /i/ and /ɪ/ contrast with regards to tongue root position. A similar study of Akan shows that vowel height also gets incorporated into the contrast (Ladefoged and Maddieson 1990). While many languages are noted for having TR contrasts, imaging data are available for only a small subset. Gua, a Kwa language from the Niger Congo family spoken in coastal Ghana (Simons and Fennig 2017, Yeboah-Obiri 2013), is a critically under-documented language which contains TR contrasts in all high and mid vowels (Advanced TR: /i e o u/; Retracted TR: /ɪ ɛ a ɔ ʊ/). Acoustic analysis and articulatory data from 3D ultrasound recordings reveal that RTR vowels show a variety of deformations of the tongue surface, depending on the vowel. However, these deformations are linked by the mechanics of tongue root retraction. Also, images reveal that differences in tongue height in addition to TR advancement are often present.

2pSC20. Phonetic realizations of the post-consonantal liquid in North and South Korean dialects. Suyeon Yun and Yoonjung Kang (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, Humanities Wing, H427, Toronto, ON M1C 1A4, Canada, suyeon.yun@utoronto.ca)

This study presents large-scale production data of consonant-liquid sequences in North and South Korean dialects. It is known that in South Korean, the liquid /L/ is not allowed in post-consonantal onset position and is nasalized not only after a nasal (e.g., /kɨmLi/ → [kɨmni] "interest") but after a stop, involving the nasalization of the stop (e.g., /sʌpLi/ → [sʌmni] "providence"). We investigate whether this holds (i) for Seoul Korean speakers who become familiar with the onset liquid through exposure to English and (ii) for North Korean speakers who retain the onset liquid in their dialect. Thirty five North Korean defectors speaking Northern Hamkyeong dialect and 20 Seoul Korean speakers read 236 Korean words including a nasal-liquid (/mL, nL, ŋL/) or a stop-liquid sequence (/pL, kL/). Acoustic measurements for the sequences include formant frequencies, high/low frequency energy ratios, and presence of closure and release. Results show that the most common pattern is the nasalized output, with speaker and lexical variation. The post-consonantal /L/ was also realized as the lateral [l] or the tap [ɾ], depending on the speaker's age (old vs. young) and the dialect (north vs. south).

2pSC21. Oral and nasal vowels effects on subglottal pressure. Didier Demolin, Roland Trouville, Ruolan Wang, and Rosario Signorello (Laboratoire de Phonétique et Phonologie, Université Sorbonne Nouvelle, 19 rue des Bernardins, Paris 75005, France, ddemolin@univ-paris3.fr)

This paper investigates the effect of oral and nasal vowels on subglottal pressure. The vowels /i, e, a, o, u, ø, œ̃, õ/ were produced on a specific tone by two Belgian French speakers (male and female). The tone frequency was given to the speakers while producing the vowels through a set of headphones connected to a synthesizer. Three frequencies were given at comparable intensities (male speaker: A, C, E; female speaker: C, F, A). Both speakers produced the set of vowels in a mask, with the mouth at a quasi-constant distance from the microphone. In addition to the acoustic signal, the subglottal pressure was recorded by tracheal puncture with a needle inserted between the cricoid cartilage and the first tracheal ring. Measurements show that both speakers produced each of the vowels with stable F0 (2 Hz difference variation from the given tone). One interesting observation is that there is a substantial difference in subglottal pressure between oral and nasal vowels. Both speakers produced nasal vowels with a lower subglottal pressure when compared to oral vowels. Mean differences between both set of vowels were quantified at 2.15 hPa. Nasal vowels were found to have lower intensity than oral vowels.

2pSC22. Cross-contextual consistency of /s/ length and spectral quality in gay men's speech. Dominique A. Bouavichith (Linguist, Univ. of Michigan, 611 Tappan St., #455C, Ann Arbor, MI 48109, dbouavichith@gmail.com)

Previous literature on gay(-sounding) speech has shown that gay men tend to produce longer /s/ tokens with higher spectral centers of gravity (Munson & Babel (2010), Levon (2006, 2007), and Linville (1998)). The existing body of work, however, has only shown this difference using word-initial, non-cluster /s/ tokens or aggregate average values across all contexts. This study specifically investigates these questions in /sC/ clusters, both word-initally and -finally. Participants produced words in a carrier phrase containing either /s/ or an /s/-stop cluster, in either word-initial or word-final position (N = 16). To control for speech rate variation, /s/ tokens were measured with respect to participants' syllable length. Preliminary findings show that gay speakers did in fact lengthen /s/ tokens in all contexts, at similar rates across cluster and singleton contexts. Significant differences between gay and straight men were found for both /s/-to-syllable ratio and spectral center of gravity at /s/ midpoints. These findings contribute to the depth of gay speech acoustics research and provide a basis for context-specific sociophonetic perceptual studies.

2pSC23. Speaking of sexuality: Analyzing [s] as an index of speaker identity in Japanese. Ryan C. Redmond (Linguist, Univ. of California Davis, 2505 5th St., Apt. 117, Davis, CA 95618, rcredmond@ucdavis.edu)

While variation in the production of the voiceless alveolar fricative [s] has been proposed as a marker of alternative sexualities in some languages, this phenomenon has yet to be tested in Japanese. In the present study, data from Japanese television shows and self-uploaded online "coming out" videos are used to sample 600 tokens of word-initial/intervocalic [s] from 20 different speakers: 10 queer men (older camp-gay "onee" celebrities, and younger Youtubers), and 10 men and women who do not identify as gay. Following a moments analysis, spectral center of gravity and skew measurements arose as the clearest delineating variables (i.e., these factors contributed to significant variation in [s] production) between the queer vs. non-queer male groups, as well as the non-queer male vs. non-queer female groups, but not the queer male vs. non-queer female groups. These findings may suggest alignment between Japanese women's language and Japanese "gay" speech, as has been noted in previous research, but two counter theories are posed. One theory notes the social benefits associated with hyperarticulation that could arise from the usage of these variant phones, while the other theory suggests that non-queer men could be the ones deviating from the norm.

2pSC24. Dialect contact and word-specific phonetics: North Koreans in Seoul. Yoonjnung Kang and Suyeon Yun (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, HW427, Toronto, ON M1C 1A4, Canada, yoonjung.kang@utoronto.ca)

This study examines the speech of North Korean refugees who currently reside in Seoul. We investigate (a) how the North Korean speakers' stops are affected by contact with Seoul Korean and (b) whether words only commonly used in the North and words newly acquired in the South are realized differently. The data were collected from 35 Hamkyeoung speakers, balanced for age (born before vs. after 1975) and time since arrival in Seoul (0–3 vs. 3–15 years). Twenty Seoul speakers also participated as a comparison group. Participants produced 78 stop-initial words with two repetitions, evenly divided into North vs. South lists, balanced for laryngeal feature, place of articulation, and the following vowel height across the lists. The results show that in line with previous literature, Hamkyeoung aspirated stops are longer in VOT than Seoul stops and Hamkyeoung lenis stops are shorter in VOT than Seoul stops. We found evidence for contact-induced phonetic drift for aspirated stops, but not for lenis stops—earlier arrivals produce aspirated stops with a shorter VOT than recent arrivals. We found no evidence of word-specific phonetic representation and stops were produced with a comparable VOT value regardless of their Northern vs. Southern origin.

2pSC25. A regression approach to vowel normalization for missing and unbalanced data. Santiago Barreda (Dept. of Linguist, UC Davis, Davis, CA 95616, sbarreda@ucdavis.edu) and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, AB, Canada)

Researchers investigating the vocalic systems of languages or dialects frequently employ normalization methods to minimize between-speaker variability in formant patterns while largely preserving dialectal, between-phoneme variability. One popular method, log-mean normalization, relies on estimating the logarithm of the geometric mean formant-frequency (G_s) produced by a speaker across their vocalic inventory, and then expressing the formant patterns produced by a speaker as deviations from this mean. However, in the face of missing or unbalanced data, the traditional approach to calculating G_s for a speaker will usually lead to biased estimates, which will produce artificial asymmetries in the normalized vowel spaces of different speakers. An alternative method is proposed for the estimation of G_s based on a linear-regression framework, which avoids the biases associated with traditional estimation of G_s when data is unbalanced. The regression method to normalization is described, and simulations are carried out to compare the accuracy of G_s estimates via regression to other estimation methods. Results indicate that the proposed method is substantially more accurate than the traditional approach to estimating G_s in the face of missing or unbalanced data.

2pSC26. Allocation of attention to real-time visual speech feedback in a digital mirror. Elizabeth D. Casserly and David E. Ballenger (Dept. of Psych., Trinity College, 300 Summit St., Hartford, CT 06106, elizabeth.casserly@trincoll.edu)

The role of sensory feedback in speech motor control is typically investigated by observing the behavioral and/or neurological response to experimental feedback perturbation. The function of feedback and its use in typical control is therefore inferred from responses under atypical conditions. The present study avoided perturbation, using eye tracking and a digital mirror (continuous video relay from a webcam) to examine how speakers attend to real-time visual feedback across normal speech and non-speech tasks. Fixation locations and durations were recorded during a period of rest prior to task onset (no-task condition), a recitation of the ABC's, a re-telling of a popular children's story (Goldilocks and the Three Bears), and parallel clear speech versions of the latter two tasks. Analysis of gaze showed that participants overall seemed to avoid fixating on their self-image, and that avoidance increased during speech tasks compared to non-speech "rest." Across the two content areas (ABC's vs. storytelling), participants' self-gaze was concentrated more heavily on speech-relevant areas during storytelling, and no differences in gaze were observed between casual and clear speech. These data suggest an increased role for sensory feedback during complex linguistic tasks, as well as indicating an overall aversion towards realistic visual self-feedback.

2pSC27. Variable vowel convergence in a novel cooperative task. Jennifer Nycz and Shannon Mooney (Dept. of Linguist, Georgetown Univ., 1437 37th St. NW, Washington, DC 20057, jn621@georgetown.edu)

Phonetic convergence is partly automatic, yet mediated by linguistic and attitudinal factors; salient social identity can suppress convergence or lead to divergence (Babel 2010). We assessed convergence across vowels among 12 pairs of speakers using a novel task that minimizes the salience of identity. Each person separately read aloud a 45-item word list, once before the main task and again afterwards. Between readings, the pair played a version of the game Taboo, in which players take turns attempting to elicit specific words from their partner while avoiding forbidden words. Lobanov-normalized formant values were extracted from the word lists. A convergence measure was calculated for each word for each pair, by subtracting the Euclidean distance between the speakers' vowels in that word in post-game lists from that of the pre-game lists. Mixed-effects models of convergence were fit, with random effects for word and pair and fixed effects for vowel and pre-game distance. Greater initial distance was associated with greater convergence. /i/ and /a/ converged most consistently, while diphthongs typically diverged. This suggests that listeners focus on point vowels to model an interlocutor's vowel space, facilitating convergence in these vowels, and that even when identity is not salient, divergence may occur.

2pSC28. Exploring the source of short-term variations in respiratory data. Susanne Fuchs (Leibniz Ctr. for General Linguist, Berlin, Germany) and Laura L. Koenig (Adelphi Univ. and Haskins Labs., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu)

Past studies of speech breathing have observed short-term variations in the respiratory signal. In early kinematic work, Stetson observed "ripples" on the breathing signals, which he interpreted to represent chest pulses for each syllable. Ladefoged and coauthors subsequently reported that brief excursions in respiratory data corresponded closely in time to stressed syllables. Segmental characteristics may also impact respiratory signals, however. In particular, voiceless obstruents have been associated with short-term decreases in breathing data, presumably reflecting rapid airflow venting through an open glottis. Indeed, much of Stetson's speech material consisted of repeated CV syllables with voiceless stop onsets. This study revisits the degree to which voiceless consonants correspond to negative excursions in respiratory signals obtained from several speakers. We collected acoustic data as well as intraoral pressure (to infer subglottal pressure non-invasively) and used inductance plethysmography to obtain displacements of the rib cage and abdomen. We use acoustic and intraoral pressure signals to locate voiceless segments as well as nasal consonants in stressed syllables, and assess the characteristics of the respiratory signals in these regions. The nasals provide a control condition for testing whether voiceless sounds have unique effects on the respiratory data.

2pSC29. Vowel contrasts relative to schwa: Effects of utterance-level fundamental frequency. Christina Kuo (Commun. Sci. and Disord., James Madison Univ., MSC4304, 235 Martin Luther King Jr. Way, Harrisonburg, VA 22807, kuocx@jmu.edu)

The purpose of this study is to characterize the potential effects of utterance-level fundamental frequency (F0) on the acoustic contrasts of vowels as expressed relative to the mid-central vowel schwa. This is a follow-up to a previous study [Kuo, *J. Acoust. Soc. Am.* **141**, 3840 (2017)] based on a hypothesis of schwa as a speaker-defined reference for vowel contrasts. The hypothesized reference schwa is made up of the averaged first and second formant (F1 and F2) frequencies from many tokens of schwa produced by a given speaker. Motivated by the potential interactions among F0, spectral sampling, and articulation, the present study evaluates the effects, if any, of utterance-level F0 on vowel contrasts relative to schwa. Utterance-level F0 is measured for breath groups at the sentence level. Vowel contrasts are expressed as the Euclidean distances between vowels and the reference schwa in the F1-F2 space. Specifically, two questions are of interest. First, are the Euclidean distances between vowels and the reference schwa impacted by the utterance-level F0? Second, does utterance-level F0 affect the variability of schwa productions, and thus influencing the makeup of the reference schwa? Findings will be discussed within the framework of the acoustic theory of speech production.

2pSC30. Pause postures in American English. Jelena Krivokapic, Will Styler (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220, jelenak@umich.edu), Ben Parrell (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, Newark, DE), and Jiseung Kim (Linguist, Univ. of Michigan, Ann Arbor, MI)

Recent work examining articulation during pauses has found that articulatory patterns distinguish grammatical from ungrammatical pauses (Ramanarayanan *et al.* 2009). For Greek, Katsika *et al.* (2014) have identified pause postures (specific configurations of the vocal tract at prosodic boundaries) which are triggered by π-gestures with high activation levels and consequently occur at strong prosodic boundaries. This study investigates pause postures in American English, specifically, whether they occur, and how they are coordinated with other events at prosodic boundaries. In an electromagnetic articulometry (EMA) study, seven speakers produced 7-8 repetitions of 42 types of sentences varying in linguistic structure (stress, boundary, phrasing, and sentence type). Results from two speakers analyzed to date indicate that pause postures exist but that their presence might be speaker dependent. Analyses of gesture lags indicate a stable relationship between the boundary tone and pause posture, that the boundary tone gesture starts later when stress is on the second syllable, and that the π-gesture is shifted towards the stressed syllable, parallel to findings in Katsika *et al.* (2014). We discuss these results in relation to models of prosodic structure and to speech planning processes. [Work supported by NIH and NSF.]

2pSC31. Tracking developmental changes in articulatory strategy during childhood. Tanner Sorensen, Asterios Toutios, Louis Goldstein, and Shrikanth S. Narayanan (Univ. of Southern California, 3740 Mcclintock Ave., Los Angeles, CA 90089, tannersorensen9991@gmail.com)

During development, children learn how to coordinate movements of the speech articulators in order to optimally achieve motor goals. It has been shown that variability in these coordinative patterns, or articulatory strategies, decreases over the course of childhood before ultimately stabilizing at adult-like levels. For example, the jaw becomes more tightly coordinated with the tongue and lips. Recent advances in real-time magnetic resonance imaging (rt-MRI) and analysis provide a means to characterize such articulatory strategies by quantifying how much the jaw, tongue, lips, velum, and pharynx contribute to constrictions of the vocal tract during speech. The articulators are segmented in reconstructed rt-MRI and constriction degrees are measured as the linear distance between opposing structures (e.g., tongue and palate). Change in constriction degree over time is decomposed into articulator contributions to characterize articulatory strategy. In this pilot study, we obtain quantitative biomarkers of articulatory strategies from a 10-year-old participant and compare them against those of 8 healthy adult participants. The study quantifies the difference between child and adult articulatory strategies in terms of how much each articulator contributes to constrictions of the vocal tract during speech and indicates how the articulator movements are coordinated with each other in time.

2pSC32. Production of stop consonants in 3-to-6-year-old Mandarin-speaking children. Jing Yang (Commun. Sci. and Disord., Univ. of Central Arkansas, CSD/ Speech Lang. Hearing Ctr., 201 Donaghey Ave., UCA Box 4985, Conway, AR 72035, jyang@uca.edu)

The present study compared the temporal measurements of stop consonants in 29 3-to-6-year-old Mandarin-speaking children and 12 Mandarin-speaking adults. Each participant produced 18 Mandarin disyllabic words which contained six stops /p, p^h, t, t^h, k, k^h/ each followed by three vowels / a, i, u/, respectively. All stop consonants were located at the word-initial position in the first syllable. The temporal measurements of VOT, overall burst duration, average duration per burst, the number of burst and VOT-lag duration were obtained. The results showed that Mandarin-speaking children in this age range produced all six stops, especially aspirated stops, with longer VOT means than the adults. The older children produced even longer VOTs than the younger children, which evidenced the overshooting pattern of VOT values in children. Although adult-like short-lag VOTs for unaspirated stops were achieved in all children, the long-lag VOTs for aspirated stops were widespread in the younger group and gradually developed to a concentrated distribution in the older children. Further examination of the burst and VOT-lag revealed that these children tended to produce shorter average duration per burst and longer VOT-lag than the adults for both unaspirated and aspirated stops. These results indicate that children in this age range may not have developed adult-like laryngeal-oral timing pattern and airflow control for stop production.

2pSC33. Practicing English medical word pronunciation with a communication robot and a tablet by Japanese medical students. Yoko Sakamoto and Nobuhiro Sakata (Premedical Sci., Dokkyo Medical Univ., 880 Kitakobayashi, Mibu, Shimotsugagun, Tochigi 3210293, Japan, y-saka@dokkyomed.ac.jp)

The aim of this study was to investigate which tool can be more effective way to practice English medical word pronunciation for Japanese medical students: a communication robot or a tablet. The subjects were 8 Japanese medical freshman students. The target words were 10 medical words related to an infectious disease. In the training, the subjects learned the meaning of the word in Japanese and then they practiced the pronunciation in English. Four subjects followed the order; Pretest → Robot → Posttest1 → Tablet → Posttest 2, while the remaining 4 subjects were counterbalanced; Pretest → Tablet → Posttest 1 → Robot → Posttest 2. All subjects answered the questionnaire after Posttest 2. The process was video-recorded and the waveform was analyzed. The results showed that the postures and voices of the subjects were better with a communication robot than with a tablet. The pronunciation practiced with a communication robot had improved with a fewer trial. The subjects wrote in the questionnaire that they could practice pronunciation with a communication robot aiming to be able to talk with a person in the future. Thus, compared to a tablet, a communication robot can be an effective tool to practice English medical words.

2pSC34. Tongue position as an articulatory property of voicing in Brazilian Portuguese and Thai. Suzy Ahn (Dept. of Linguist., New York Univ., 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu)

Articulatory adjustments are often necessary to ensure that closure voicing will be present in stops (Rothenberg 1968). One common adjustment is to enlarge the supralaryngeal cavity volume via tongue root advancement (Westbury 1983). This study uses ultrasound to examine tongue positioning during Brazilian Portuguese and Thai stops. Portuguese has a two-way laryngeal contrast: voiced and voiceless (unaspirated), and Thai has a three-way contrast: voiced, voiceless unaspirated, and voiceless aspirated. Eight native speakers of each language recorded phrase-initial stops followed by /a/. Results show a clear distinction in tongue position between voiced and voiceless unaspirated stops in both languages. Tongue root is more advanced for voiced compared to voiceless unaspirated in alveolar and velar stops. For labial stops, Thai speakers lower the tongue front (which is another cavity enlargement maneuver) whereas Portuguese speakers advance their tongue root for voiced stops. The results suggest that speakers of both languages employ tongue positioning for voicing during closure. How the tongue position is operationalized may be language-specific as long as voiced stops have a larger cavity volume. On the other hand, the role of tongue position for aspiration is less clear since in Thai, alveolar stops and velar stops show the opposite pattern when voiceless unaspirated and aspirated stops are compared.

2pSC35. Nasalization as a correlate of word-final voicing in English. Suzy Ahn (Dept. of Linguist., New York Univ, 10 Washington Pl., New York, NY 10003, suzy.ahn@nyu.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

Maintenance of laryngeal voicing during closure of word-final stops is believed to be articulatory challenging (Westbury & Keating, 1986). Additional articulatory maneuvers may help sustain voicing during final closure, and one such maneuver is pre-nasalization (Rothenberg, 1968). It has been demonstrated that pre-nasalization is associated with voicing in utterance-initial stops in French, Spanish, and to a lesser extent, in English where voiced stops are often realized without voicing during closure (Solé, 2011). Laryngeal contrast is typically maintained between word-final voiced and voiceless stops in English, and voicing is often found during closure. Therefore, articulatory maneuvers supporting the phonetic realization of this contrast are particularly important in final position. The proposed study examines the degree of nasalization in the vowel immediately preceding word-final stops in monosyllabic real words of American English (N = 18, monolingual speakers of Mid-western dialect). Preliminary examination of word-final stops in English indicates that pre-nasalization occurs for word-final voiced stops as well. Moreover, the presence of nasalization is predicted to correlate with the amount/duration of voicing during the stop closure of final consonant. The results will contribute to our understanding of the phonetics of word-final laryngeal contrast and potentially expand the inventory of secondary perceptual cues to word-final voicing.

2pSC36. Onset f0 as a correlate of voicing in Marathi. Olga Dmitrieva (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu) and Indranil Dutta (The English and Foreign Lang. Univ., Hyderabad, India)

Onset f0 (the fundamental frequency at the onset of vowel immediately following the consonant) has been shown to correlate with voicing in a variety of languages with two-way voicing contrasts in stop consonants, such as the voice-type contrast (between prevoiced and voiceless unaspirated stops) or the aspiration contrast (between voiceless unaspirated and voiceless aspirated stops). Onset f0 is typically higher after phonologically voiceless than after voiced stops. With some notable exceptions, onset f0 data from languages with three-way and four-way voicing contrasts is relatively scarce. The present study explores onset f0 as a correlate of voicing in Marathi, an Indo-Aryan language with a four-way voicing contrast which includes both aspirated and unaspirated voiced and voiceless stops. Results demonstrate that onset f0 covaries with both laryngeal voicing and aspiration in Marathi: f0 is significantly lower for both sets of voiced than voiceless stops. Onset f0 is also significantly lower for both sets of aspirated than unaspirated stops but the effect of voicing is greater in magnitude and more consistent across individuals than the effect of aspiration. Combined with previous findings, the data suggest that onset f0 covariation with laryngeal voicing is more universal across languages than its covariation with aspiration.

2pSC37. Characteristics of saliva swallowing in the reading of prepared texts. Kuniko Kakita (Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, kakita@pu-toyama.ac.jp) and Shizuo Hiki (Waseda Univ., Tokorozawa, Japan)

This study examines the nature of saliva swallowing during speech production. Speakers read two contrasting types of texts, a "paragraph" and a "list." The "paragraph" was a Japanese version of "The North Wind and the Sun," consisting of four sub-paragraphs. The "list" was a simple repetition of quasi-identical short sentences, without any paragraph construction. Simultaneous recordings of speech sounds (via microphone), swallow sounds (via laryngeal contact microphone), and respiratory movements of the thorax and abdomen (via respiratory belt transducers) were obtained and analyzed, especially in relation to sentence-final pauses. Preliminary results indicate that (1) swallowing tends to occur roughly in the middle of a pause and is followed by inspiration for the upcoming utterance, (2) inspiration is generally deeper when preceded by swallowing, (3) swallowing adds approximately one second to a pause, and (4) in paragraph reading, swallowing is more likely to occur during a pause between two sub-paragraphs, i.e., at a topic transition, whereas in list reading, swallowing reflects speakers' physiological needs more directly.

2pSC38. Novel imaging tools for supporting the teaching of singing and spoken performance. Reed Blaylock and Shrikanth S. Narayanan (Univ. of Southern California, 1150 W 29th St. Apt. 4, Los Angeles, CA 90007, reed.blaylock@gmail.com)

Recently, singers and singing instructors have begun to use real-time magnetic resonance imaging (rtMRI) videos of speech and singing movements as a pedagogical tool. Singers use videos of vocal tract movements to learn about the movements of speech, and are then able to contrast those movements with their ideal singing productions. These validated images can now replace the previous "practice-of-speculation" regarding the articulatory postures desirable during singing. While rtMR videos are most often used by professional instructors and aspiring-professional singers, these videos may soon commonly be used to assist amateur choruses as well, which often struggle to produce unified vowel qualities. Beatboxing is a related and unexplored pedagogical domain, yet beatboxing was one of the earliest spoken performance styles acquired at USC with real-time MRI. These videos, together with USC's publicly available inventory of rtMR videos demonstrating the sounds of the world's languages, provide a rich resource for the extreme and varied percussive sounds used by beatboxers. In sum, students of singing and spoken language performance have a brand new asset available to them in the form of dynamic real-time MR videos of the moving vocal tract. [NIH R01DC007124.]

Session 2pSP

Signal Processing in Acoustics and Underwater Acoustics: Detection, Classification, Localization, and Tracking (DCLT) using Acoustics (and Perhaps Other Sensing Modalities) II

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Contributed Papers

1:00

2pSP1. Simulation of passive source localization in near-Arctic conditions using frequency-difference matched field processing. David J. Geroski (Appl. Phys., Univ. of Michigan – Ann Arbor, Randall Lab., 450 Church St., Ann Arbor, MI 48109, geroskdj@umich.edu) and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, Ann Arbor, MI)

Matched Field Processing (MFP) is a well-known technique for passive source localization in complex acoustic environments. It involves correlating array-recorded acoustic fields with replica fields calculated using a model of the acoustic environment, and works well is known environments. However, acoustic signals often depend on unknown environmental details which are not included in the replica-field calculations. The severity of this mismatch increases with frequency and source-array range, and it can cause MFP to fail in the signal band at relevant ranges in the arctic ocean. A proposed remedy to this problem is to utilize the frequency-difference autoproduct of the measured acoustic field instead of the acoustic field itself, and to perform the replica calculations at the difference frequency. The simulated performance of frequency difference MFP is shown for a generic near-Arctic environment at signal frequencies of 200 to 300 Hz, and ranges of 30 to 300 km. Here, mismatch between in the acoustic and replica field is modeled by imposing a range-dependent altimetry and surface sound speed profile to simulate surface ice, and by random time delays applied to each ray-path between the source and each receiver to simulate refractive index fluctuations in the ocean. [Sponsored by ONR.]

1:15

2pSP2. Passive acoustic localization and tracking using synthetic data for the Northern Gulf of Mexico. Britt J. Aguda, Kirk D. Bienvenu (Physics, Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, bjaguda1@uno.edu), Bradley J. Sciacca (Physics, Univ. of New Orleans, Harvey, LA), Joshua Veillon, SydniCherise O. Austin, and Juliette W. Ioup (Physics, Univ. of New Orleans, New Orleans, LA)

Passive acoustic localization and tracking of marine mammals shows potential for future acoustic monitoring efforts. The Littoral Acoustic Demonstration Center—Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015 using Environmental Acoustic Recording Systems (EARS) buoys, returning to sites previously surveyed by LADC. The localization and tracking method developed for the EARS hydrophones uses Monte-Carlo based simulations based on this data. Synthetic whale clicks at random times along a sinusoidal path were propagated to the EARS receivers and the location of the synthetic source was tracked. Several whale clicks in a series determine the most probable path of the animal. The localization/tracking method and relative localization are adaptable for variables such as site location, mooring locations, number of moorings, depth, etc. Localization of two synthetic moving whales in three-dimensional space will be demonstrated. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

1:30

2pSP3. Future deployment corrections for passive acoustic localization and tracking using real data for the Northern Gulf of Mexico. Kirk D. Bienvenu, Britt J. Aguda (Physics, Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, kdbienv1@uno.edu), Bradley J. Sciacca (Physics, Univ. of New Orleans, Harvey, LA), Joshua Veillon, SydniCherise O. Austin, and Juliette W. Ioup (Physics, Univ. of New Orleans, New Orleans, LA)

The Littoral Acoustic Demonstration Center–Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015 using Environmental Acoustic Recording Systems (EARS) buoys. A localization and tracking method developed for the EARS hydrophones uses Monte-Carlo based simulations based on these data but was only tested using synthetic whale clicks at random times along a sinusoidal path. The location of the synthetic source was tracked. Real data obtained in 2015 could not be used by the method developed due to excessive clock drift between moorings with no way to correct the phase differences. It will be shown, using the geometry of the site and acoustic theory, how a simple, low powered, periodic acoustic ping can aid in correcting the phase difference of the clocks after significant time has passed. [This research was made possible by a grant from The Gulf of Mexico Research Initiative. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

1:45

2pSP4. Spatial-division-multiplexing in time-variant and multi-path channel based on vector adaptive time reversal technique. Dian Lu, Xueli Sheng, Yali Shi, Hanjun Yu, and Longxiang Guo (Underwater Acoust. Eng. College, Harbin Eng. Univ., No.145, Nantong Rd., Nangang District, Harbin, Heilongjiang Province 150001, China, bryce_era@sina.com)

In the field of multistatic remote detection for underwater target, as a time-variant and multi-path channel interfered by environmental noise, complex underwater acoustic environment limits the spatial share of channel and poses a great challenge against spatial-division-multiplexing (SDM) in multistatic detection. In this paper, a kind of method of spatial-division-multiplexing(SDM) based on vector adaptive time-reversal technique is

proposed. By use of time-reversal technique, multi-path structure of channel is efficiently restrained; Meanwhile, with adaptive filtering applied, time-variant characteristic of channel is effectively suppressed. Moreover, with single vector hydrophone, spatial focus of target echoes and noise interference suppression is accomplished by spatial filtering. Finally, excellent results are performed in SDM of multistatic detection, supported by the experimental simulation.

2:00

2pSP5. An improved target detection and azimuth angle estimation method using a single acoustic vector sensor. Lin Ma, Anbang Zhao (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, HLJ 150001, China, malin@hrbeu.edu.cn), Juan Hui, Caigao Zeng, and Xuejie Bi (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

In this paper, an improved underwater acoustic target detection and azimuth angle estimation method using a single acoustic vector sensor (AVS) is proposed based on time-reversal and convolution theory. The proposed method can be applied in the active and the passive sonar detection system. According to the conventional detection and estimation method based on complex acoustic intensity measurement, the mathematical and physical model of this proposed method is described in detail. Computer modelling and simulation is applied to demonstrate the proposed method's effectiveness. In order to further verify the practical application performance of the proposed method, the research group carried out the open lake experiments. The computer simulation and open lake experiments results indicate that this method can realize the azimuth angle estimation with high precision by using only a single AVS. Compared with these conventional methods, the proposed method achieves better detection and estimation performance.

2:15

2pSP6. Detection performance analysis of product processing of colinear arrays. Kaushallya Adhikari (Louisiana Tech Univ., 600 Dan Reneau Dr., Ruston, LA 71270, kaushallyaadhikari@gmail.com) and John R. Buck (Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

Non-uniform linear arrays (NULAs) often achieve better resolution than standard uniform linear arrays (ULAs) with equal numbers of sensors. However, conventional beamforming (CBF) of an NULA received signal leads to the same detection statistic PDF as a ULA with the equal number of sensors, undercutting NULA's improved resolution. Nested and coprime arrays partition an NULA into two subarrays and multiply the subarrays' CBF outputs. This research compares this product processor's detection performance against a CBF detector with an equal number of sensors for a narrowband Gaussian signal in spatially white additive Gaussian noise. The product processor's detection PDF is a scaled product of the detection statistic with modified Bessel functions. Receiver operation characteristics (ROC) curves illustrate that the product processor's performance is inferior to the CBF detector with an equal number of sensors. The detection performance of a product processor matches the CBF detector only for high SNRs and large numbers of sensors. However, in the presence of interferers, the product processor for coprime arrays can outperform both the CBF detector and product processing nested arrays with an equal number of sensors. [Work supported by ONR.]

2:30

2pSP7. Double-difference tracking of bowhead whales using unsynchronized directional acoustic recorders in the Beaufort Sea. Ludovic Tenorio-Hallé (Scripps Inst. of Oceanogr., 1044 Loring St., San Diego, CA 92109, ludovictenorio@gmail.com), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), and Katherine H. Kim (Greeneridge Sci., Inc., Santa Barbara, CA)

Passive acoustic monitoring has become a standard method for detecting bowhead whale (*Balaena mysticetus*) activity in Arctic waters. Between 2007 and 2014, over 40 directional acoustic recorders, known as DASARs, were deployed in the Beaufort Sea during the bowhead whale migration season. Individual DASARs can estimate azimuth, allowing calls to be localized by triangulation using multiple DASARs. However, these bearings are subject to calibration biases, and individual recorders were not precisely time-synchronized, making relative time-of-arrival information unusable for standard localization purposes. Double-difference methods have previously been applied in seismology to obtain high-precision relative positions of earthquakes by measuring changes in relative travel-times between multiple events over widely distributed seismic sensors. Here, the double-difference method is applied to detected bowhead whale calls in order to improve the relative localization resolution. The approach uses changes in both relative call travel-times and bearings, detected at multiple DASARs, to determine high-precision relative locations of these calls despite the presence of systemic timing and bearing errors in the measurements. The resulting positions allow tracking of individual whales, which may provide insight into the function of these calls.

2:45

2pSP8. Research on passive detection parameter measurement technology based on single vector sensor. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn)

Passive detection as a major function of the sonar system, over the years has been the work of waterborne workers to solve the problem. This paper mainly talks about the measurement technology of passive detection parameters based on single vector sensor. The problem of low frequency line spectrum signal extracted from ship radiated noise is studied, and the azimuth sequence and frequency sequence are obtained by estimating and analyzing the low frequency line spectrum signal. Ship radiated noise contains a lot of target information, so the extraction of the line noise of the ship's radiated noise is of great significance in the measurement of passive detection parameters. The radiation noise of the ship is a broadband spectrum modulated by the spectral signal envelope. The average azimuth and complex intensifier are used to estimate and analyze the target azimuth, and the estimated angle is statistically processed by histogram and weighted histogram. For the frequency estimation technique, the adaptive frequency estimator based on the adaptive notch filter and the adaptive LMS (Least Mean Square) algorithm is chosen to study the target signal frequency. Through the simulation experiment, the azimuth estimation technique and the adaptive frequency estimation technique are feasible. By estimating the azimuth sequence and frequency sequence of the target radiation signal, the relevant information of the target can be obtained, so it is of great significance in the research of underwater acoustic detection technology.

3:00–3:15 Break

3:15

2pSP9. Acoustic localization of distributed coherent and incoherent sources using SEMWAN with subarray smoothing. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

Accurate estimation of the location and level of remote acoustic sources from recorded acoustic signals is attractive in many applications. In wind tunnel tests, where noise sources are commonly distributed and incoherent, high resolution array signal processing techniques like the spectral estimation method with additive noise (SEMWAN) have been useful for source localization when background noise measurements are available. However, for many continuously distributed systems, such as a simple vibrating plate, the assumption of incoherent sources is incorrect, and techniques like SEMWAN may yield spurious results. In this presentation, results are reported for the use of SEMWAN alongside a subarray smoothing technique to formulate the coherent source localization problem as an incoherent source localization problem. Simulations comparing localization performance for distributed coherent and incoherent sources are shown. Results from a proof-of-concept experiment using multiple sources and a 15-element linear receiver array are also evaluated against simulation. Performance comparisons are made between SEMWAN, MUSIC, and conventional beamforming techniques in addition to showing the effects of subarray smoothing. [Sponsored by NAVSEA through the NEEC and by the U.S. DoD through an NDSEG Fellowship.]

3:30

2pSP10. Experimental investigation of the unit circle minimum variance distortionless response adaptive beamformer. Matthew Tidwell and John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, mtidwell@umassd.edu)

Source masking occurs when loud interferers produce high side lobes in the conventional beamformer (CBF) scanned response, obscuring the true direction-of-arrival of the desired signal. The minimum variance distortionless response (MVDR) adaptive beamformer (ABF) places deep beampattern notches near interferer directions to suppress their power in the ABF output while maintaining unity gain in the look direction. In practice, the sample matrix inversion (SMI) MVDR ABF replaces the ensemble covariance matrix with the sample covariance matrix (SCM) when computing the array weights. Estimation of the SCM in snapshot-limited and snapshot-deficient scenarios causes perturbations of the array polynomial zeros from the unit circle. The unit circle (UC) MVDR ABF projects the array polynomial zeros radially back to the unit circle, producing deeper notches and reduced sidelobe levels [Tuladhar and Buck, ICASSP 2015]. Preliminary experiments in an outdoor, free-field environment with a 21-microphone uniform linear array at a design frequency of 2.3 kHz found the UC MVDR ABF consistently outperformed both the conventional and SMI MVDR beamformers in interferer suppression when limited to 21 snapshots (one snapshot/sensor). Additionally, the UC MVDR beamformer averaged 18 dB better white noise gain than the SMI beamformer across six independent trials. [Research funded by ONR.]

3:45

2pSP11. Automated passive acoustic detection, classification, localization, and tracking methods applied to recorded data from the Pacific Missile Range Facility, Kauai, Hawaii. Stephen W. Martin (Environ. Program, National Marine Mammal Foundation, 1929 Cherrywood St., Vista, CA 92081, martin.steve.w@gmail.com), Brian Matsuyama (Environ. Program, National Marine Mammal Foundation, San Diego, CA), Tyler A. Helble (Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA), Cameron R. Martin (Environ. Program, National Marine Mammal Foundation, San Diego, CA), E. E. Henderson (Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA), and Gabriela C. Alongi (Environ. Program, National Marine Mammal Foundation, San Diego, CA)

Automated methods will be reviewed for performing passive acoustic detection, classification, localization, and tracking of some marine mammal species and man-made sources. The methods have been applied to recorded hydrophone data from a large aperture seafloor array at the Pacific Missile Range Facility (PMRF) at Kauai, Hawaii, with some of the methods currently implemented in a real-time system at PMRF. The process consists of custom software both in C++ and Matlab in a 3 or 4-step process. Automated detections of various sounds in specific frequency bands are first performed. In some cases, a classification stage is also performed. The third stage involves model-based localization of the detections or classifications. The fourth stage converts the localizations into individual source tracks. Source tracks are currently generated for fin, sei, Bryde's, humpback, and sperm whales and mid-frequency active sonar (MFAS) transmissions. Performing this process on marine mammals allows information regarding the movement patterns of the whales while calling, as well as information on the species' calls (e.g., call rate, frequencies, durations, estimated source levels). By performing similar processes on man-made sources, it is possible to determine some marine mammal responses from proximity of Navy vessels and mid-frequency active sonar sources.

4:00

2pSP12. Detection and localization of weak impulses and continuous sources within the urban acoustic environment. Richard Haskins, Christopher Simpson, and Mihan H. McKenna (usace, ERDC, 3909 Halls Ferry Rd., Vicksburg, MS 39180, richard.w.haskins@usace.army.mil)

Signal detection and localization with large aperture acoustic arrays in urban environments can be inherently difficult. This poster presents some preliminary approaches under investigation for; determining the direction of arrival of continuous sources that are changing in frequency, detecting and localizing low signal to noise ratio impulses in the presence of coherent background noise, and a fast mapping approach for converting array element delays to direction of arrival using a neural network fitting function. For uniform circular array geometries with a central reference array element, the direction of arrival of sources with changing frequency can be visualized by applying the Hilbert transform and analyzing the relative phase angle rate of change of the outer array elements. Next, various computational approaches for low level impulse detection in the presence of coherent acoustical noise are presented. Last, a neural network fitting function is discussed that performs array specific delay mapping to direction of arrival with high computational efficiency and trained noise immunity.

4:15

2pSP13. Small aperture acoustic methods to locate transients. Michael V. Scanlon (US Army Res. Lab., Acoust. & EM Sensing Branch, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Acoustic microphone arrays have been used in battlefield environments to detect and locate continuous wave targets like helicopters and ground vehicles, and for transient events like gunshots, mortars, rockets and explosions. Depending on the application, source bandwidth, propagation distance, and required localization accuracy, these arrays can vary their sensor separation greatly. Unattended ground sensors, manned- and unmanned-platforms, and Soldier-worn systems can all benefit from very small apertures if performance can be maintained. A commercially available acoustic particle velocity sensor, approximately one-half an inch in diameter, showed good transient localization at a field experiment. Data will also be presented from a small cluster-array of cardioid microphones.

4:30

2pSP14. Evaluating direction of arrival uncertainty in undersea canyons with internal tides. Timothy F. Duda (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Bruce Cornuelle (Scripps Inst. of Oceanogr., La Jolla, CA), Ying-Tsong Lin, Arthur Newhall, and Weifeng G. Zhang (Woods Hole Oceanographic Inst., Woods Hole, MA)

The slopes of undersea canyon regions impart complexity to underwater acoustic fields for two reasons: intricate patterns of reflection from the seabed, and sound-speed anomalies from internal tides generated at the slopes. Both may spread the horizontal directional spectrum of sound from "line of sight" to a source. The directional spectrum is tied to the covariance matrix of the field, a fundamental quantity that can be measured or modeled. Here, sound-field horizontal-lag spatial covariance matrices and other derived quantities are generated from time-stepped 3D parabolic equation acoustic simulations made using sound-speed fields from ocean models. The covariance matrices are then inserted into the direction-of-arrival (DOA) estimation problem. Analysis of DOA estimates and error bounds is done for a conventional beamformer and for a Gauss-Markov inverse-based beamformer for a variety of signal-to-noise ratios. The method treats non-line of sight acoustic energy as a form of noise. At low signal-to-noise ratio, the Gauss-Markov estimator can perform better that the other. This analysis of field variability allows performance degradation caused by evolving ocean structures to be directly compared to other detrimental influences such as excess noise and array deformation. Detection, localization, and tracking are affected by the processes examined here.

4:45

2pSP15. Extrapolated open spherical microphone arrays beamforming for acoustic source localization. Boquan Yang, Shengguo Shi, Ying Li, Lanyue Zhang, and Chao Wang (Harbin Eng. Univ., NanTong Str NanGang Dist., Harbin, HeiLongJiang 150001, China, ybq416751946@163.com)

Spherical microphone arrays have become a particular tool for analyzing the spatial sound field, especially in source localization and identification. Spherical Harmonic Beamforming (SHB) is a fundamental algorithm with the spherical microphone arrays in aeroacoustics. However it presents some intrinsic limitations, specific poor resolution and sidelobe suppression capability. This paper aims to overcoming this limitations by employing Functional Beamforming and Beamforming Regularization Matrix. Functional beamforming has a narrower the mainlobe width and lower sidelobes by

increasing the order of matrix functions. To improve the performance of algorithm at low frequency, this paper uses an extrapolated spherical arrays that is a larger radius of the spherical arrays through the Beamforming Regularization Matrix algorithm which can improve the spatial resolution. The results of numerical simulations and experiments show that the proposed method can remarkably suppress beamforming sidelobes, improve dynamic range of the system, and obtain a higher array gain, which leads to the accurate identification the location of sound sources.

TUESDAY AFTERNOON, 5 DECEMBER 2017 SALON F/G/H, 1:00 P.M. TO 4:45 P.M.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques II

David P. Knobles, Cochair
KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Invited Papers

1:00

2pUW1. In situ measurements of sediment sound speed and attenuation at the Seabed Characterization Experiment site in the frequency band of 0.5–10 kHz. Jie Yang (Acoust. Dept., APL-UW, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

The Seabed Characterization Experiment, sponsored by the Office of Naval Research, was carried out 5 March–10 April, 2017 (SCE17) on the New England Mud Patch, approximately 100 km south of Martha's Vineyard. The main SCE17 experimental site covers an area of 15 km x 30 km with water depths of 75–80 m. The Sediment Acoustic-speed Measurement System (SAMS) is designed to measure sediment sound speed and attenuation simultaneously over the surficial 3 m of sediments. SAMS consists of ten fixed sources and one receiver which is driven into the seabed. The ten sources combined covers a frequency band of 0.5-10 kHz. During SCE17, SAMS was successfully deployed at 18 sites, which were chosen to measure sediment sound speed and attenuation in both the surficial mud layer and the sandy basement to support modeling effort. In addition, all 18 sites have co-located gravity and/or piston cores that were collected during the two survey cruises in 2015 and 2016. In this talk, a summary of SAMS operation during SCE17 is presented, as well as the preliminary results of sediment sound speed and attenuation and their spatial variation in the frequency band of 0.5–10 kHz. [Work supported by ONR.]

1:20

2pUW2. Geophone and tetrahedral array measurements in the New England Mud Patch and preliminary results. Gopu R. Potty, James H Miller, Poonam Aggarwal, Nipun Aggarwal (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Ying-Tsong Lin, and Arthur Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA)

Measurements of acoustic pressure and particle velocity were made during the Seabed Characterization Experiment (SCEx) in the New England Mud Patch south of Cape Cod in about 70 meters of water. The University of Rhode Island and Wood Hole Oceanographic Institution deployed the "geosled" with a four-element geophone array, a tetrahedral array of four hydrophones and several hydrophone receive units (SHRUs). In addition, a new low frequency source, interface Wave Sediment Profiler (iWaSP) was deployed to excite interface waves. The iWaSP system consists of a source to generate the interface wave and a four-element accelerometer receive array. The range between the iWASP and geosled was about 100 meters. Examples of data collected during the experiments will be presented. Preliminary results of the acoustic and particle velocity data will be discussed. Preliminary analysis of the data and approximate initial estimates of the seabed properties will be presented. [Work supported by Office of Naval Research.]

1:40

2pUW3. Preliminary analysis of geophone and tetrahedral array data from the Seabed Characterization Experiment. Poonam Aggarwal, Nipun Aggarwal, Gopu R. Potty, James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02882, poonam.aggarwal@gmail.com), Ying-Tsong Lin, and Arthur Newhall (WHOI, Woods Hole, MA)

As part of the Seabed Characterization Experiment (SCEx), the University of Rhode Island and Wood Hole Oceanographic Institution deployed the low frequency shear measurement system and the interface Wave Sediment Profiler (iWaSP) system in the New England Mud Patch south of Cape Cod in about 70 meters of water. Multiple sensors were utilized to collect data which include a geosled with a geophone array of four vertically gimbaled geophones and a tetrahedral array of four hydrophones. The iWasp was used for exciting the interface waves. The iWaSP system was deployed from the R/V Sharp and it transmitted 10 seconds chirp signals every one minute between 33 and 200 Hz. The geophone array was approximately 100 m from the iWaSP source. Detection of the chirps from geophone and tetrahedral array signals was challenging due to the presence of noise sources such as ship traffic and environmental noise. Data were explored for the presence of iWaSP chirp signals in the audio and further analyzed using techniques such as matched filtering. The preliminary results and initial hypothesis regarding the wave types corresponding to the multiple arrivals will be presented. [Work supported by Office of Naval Research.]

2:00

2pUW4. Geoacoustic inferences from seabed reflection measurements on the New England mud patch. Charles W. Holland, Chad M. Smith (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Josée Belcourt, and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of VIctoria, Victoria, BC, Canada)

The vast majority of sediment acoustics research has been aimed at understanding propagation in granular (sandy) sediments. The focus of the ONR Seabed Characterization Experiment was to improve understanding of fine-grained/cohesive sediments. A variety of measurement techniques by various researchers were conducted to that end. Here, we report on initial results from broadband wide-angle reflection measurements at two sites, one with a $\sim$2 m "mud" layer thickness and the other $\sim$11 m thick. The measured reflection coefficient exhibits features that permit estimation of geoacoustic properties including the critical angle (with a rather weak frequency dependence, 200-2500 Hz) and interference patterns in frequency-angle space (which provide information on properties in individual layers). Modeling permits insight into the information content of the data and some initial estimates of the geoacoustic properties. [This research was funded by the Office of Naval Research, Ocean Acoustics Program.]

2:20

2pUW5. Bayesian geoacoustic inversion of seabed reflection data at the New England mud patch. Josée Belcourt, Stan E. Dosso (Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Bob Wright Ctr. A405, Victoria, BC V8P 5C2, Canada, joseebelcourt@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Jan Dettmer (GeoSci. Dept., Univ. of Calgary, Calgary, AB, Canada)

This paper presents nonlinear Bayesian inversion of wide-angle seabed reflection-coefficient data for fine-grained/cohesive sediments recorded in the ONR Seabed Characterization Experiment at the New England mud patch. In particular, the inversion is applied to high-resolution broadband reflectivity data from a site with a $\sim$11-m thick mud layer. Trans-dimensional inversion, sampling over an unknown number of seabed layers, and spherical-wave reflection modeling are employed. The inversion provides maximum a posteriori parameter estimates with uncertainties quantified in terms of marginal posterior probability profiles for sound speed, density, and attenuation as a function of depth in the sediment. [The research was funded by the Office of Naval Research, Ocean Acoustics Program, and the Canadian Department of National Defence.]

2:40–3:00 Break

3:00

2pUW6. Surface wave effects on bottom geoacoustic inversions. Ying-Tsong Lin, Timothy F. Duda, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA)

Underwater sound propagation in the ocean can suffer from the scattering, focusing and defocusing effects caused by surface gravity waves. These effects can in fact also influence bottom geoacoustic inversions, which use sound pressure field measurements in the water column to estimate/infer acoustic properties of the seafloor and/or seabed. In our study presented in the paper, analytical, numerical and experimental approaches are taken to investigate how surface gravity waves can affect bottom geoacoustic inversions (the dependencies on acoustic frequency, surface wave spectrum and directivity, and acoustic waveguide parameters.) The analytical approach is based on acoustic mode theory, and the numerical approach is utilizing both the parabolic-equation (PE) model and the ray model. Experimental data were collected from the Seabed Characterization 2017 experiment conducted on the New England shelf in March and April, 2017. During the experiment, several storms passed through the experimental area, and broadband sound transmissions (500 to 1000 Hz) from fixed sources to fixed receivers have shown significant acoustic effects of surface wind waves. These sound transmissions were at 160 dB source level and had two scheduling window types, one with rapidly repeated sampling and one with less-frequent sampling for a long duration. [Work supported by the Office of Naval Research.]

3:20

2pUW7. Measuring broadband wide angle bottom loss using a hydrophone equipped underwater glider. Yong-Min Jiang (Res. Dept., NATO-STO-Ctr. for Maritime Res. & Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, yong-min.jiang@cmre.nato.int)

Seabed bottom loss is an important factor for predicting sound transmission loss in water. The NATO–STO–CMRE has been investigating novel remote sensing solutions for transforming well established bottom characterization methodologies developed for scientific studies to autonomous platforms. During ONR sponsored Seabed Characterization Experiment 2017, an omni-directional hydrophone equipped Slocum glider was adopted to measure the wide angle bottom loss of a mud patch in New England, 60 miles south of Martha's Vineyard, Massachusetts, USA. An omni-directional acoustic source was deployed over the side of R/V Endeavor to a depth of 30 meters under the sea surface. Chirp pulses ranging from 2 to 20 kHz in frequency domain were transmitted by the acoustic source. The glider was programmed to glider from and then back to the source. Broadband, wide angle bottom loss as a function of grazing angle and frequency was obtained by registering the ratios of bottom reflected and direct arrivals measured at the hydrophone on the glider over an angular range and at different 1/3 octave bands. The measurement technique, signal processing procedure and preliminary results are presented in this paper. [*Work funded by NATO-Allied Command Transformation and Office of Naval Research—Global.*]

3:40

2pUW8. Characterization of mud sediments using the frequency dependence of phase velocity and attenuation of compressional waves. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

The present paper's theory predicts phase velocity and attenuation for mud sediments that contain silt particles, and is based on the model of a suspension consisting of solid particles dispersed in a viscous liquid; the attenuation expression dates back to Lamb's Hydrodynamics and to Urick (JASA, 1948), clarified and rederived by Pierce, Siegmann, and Brown (POMA, 2017). The application to mud is based on the premise that silt particles are held in suspension by the loosely connected matrix of clay particles and that their natural oscillation frequencies are significantly less than the frequencies used in underwater acoustics. The present paper extends that theory with a fresh derivation based on concepts of matched asymptotic expansions and results in an expression for the complex wave number k as a function of the angular frequency. The results for phase velocity disagree with results published in the past by Ahuja (JASA, 1972) and Temkin (JASA, 2000). The assertion is made that the use of the theoretical predictions will enable one to deduce mud sediment properties such as porosity, fraction of volume that is suspended particles, and representative grain sizes from measurements of frequency dependence of sound speed and attenuation. [Work supported by ONR.]

Contributed Papers

4:00

2pUW9. Bayesian inference and model selection to investigate seabed shear-wave velocity profiles via interface-wave dispersion inversion. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca) and Hefeng Dong (Dept. of Electron. Systems, Norwegian Univ. of Sci. & Techonolgy, Trondheim, Norway)

This paper applies a general geoacoustic profile parameterization based on Bernstein polynomial basis functions to consider the form of seabed shear-wave velocity profiles via Bayesian inversion of interface (Scholte) wave dispersion. Theory and observations indicate that the shear-wave velocity profile of seabed sediments of uniform composition and density often corresponds (approximately) to a power law function of depth. Hence, past inversions have often been based on a power-law parameterization without considering independently if this form is actually required by the data. The Bernstein parameterization, based on a linear combination of Bernstein-polynomial basis functions with the polynomial order determined by the Bayesian information criterion, is general and allows the form of the profile to be determined by the data, rather than by a subjective prior choice. In this paper, measured-data inversions are compared for power-law, Bernstein-polynomial, and layered trans-dimensional parameterizations to investigate the shear-wave velocity profile form.

4:15

2pUW10. Applications of a suspension theory of silt particles in mud to frequency and other parameter dependencies of geoacoustic quantities. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Retired, East Sandwich, MA), Jason D. Chaytor (U.S. Geological Survey, Woods Hole, MA), Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

A recent suspension theory of marine mud [Pierce, *et al.*, POMA 29, accepted] hypothesizes that embedded silt particles are the dominant contributors to compressional wave attenuation. The approach predicts frequency intervals within which attenuation increases roughly linearly with frequency, as often assumed. These intervals depend on the measured (or assumed) mean silt particle size. This presentation investigates the influence of distributions of silt particle sizes on attenuation, including the distribution shapes obtained from data, and the intervals of linear frequency with multiple particle sizes. In addition to attenuation, the theory also provides compressional sound speed predictions. Their sensitivity to changes in measured physical parameters and frequency will be determined, and the results compared with archival data and recently analyzed SBCEXP core data along a range-dependent experimental track. Another consequence of the theory is how porosity is affected by the minimum separation distance between silt particles and including viscous boundary layers. An environmental-acoustic model of the experimental track will be constructed, based on ocean sound-speed profiles, bathymetry, analyzed core data, and attenuation from theory. This model will be used to calculate transmission loss for propagation along the track to compare with acoustic measurements from the experiment. [Work supported by ONR.]

4:30

2pUW11. Physical and acoustical properties of marine sediments with a wide particle size distribution. Justin T. Dubin, Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, justin.dubin@utexas.edu), Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Naturally occurring marine sediments can possess poorly sorted or multimodal particle size distributions. Furthermore, while well-sorted coarse grained sediments are typically characterized according to their grain size, for mixed or fine grained sediments the shape of the particles can have significant influence on the acoustical properties. In this work, particle size and morphology are investigated using scanning electron microscopy (SEM). Images of dry particles are analyzed to determine the sphericity and angularity of sand and silt grains and to estimate the shape (e.g. platelets or needles) and aspect ratio of clay particles. SEM images of wet samples are analyzed to understand the sediment microstructure, including the interaction between sand, silt, and clay particles. Additionally, the effect of the proportionally varying composition of sand, silt, and clay particles on the bulk properties of the sediment, including the porosity and compressional wave speed, will be examined using a multiphase effective medium model. These techniques are applied to sediments collected in the New England Mud Patch as part of the Seabed Characterization Experiment. [Work supported by ARL:UT IR&D and ONR.]

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, 5 December

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Studio 7
Acoustical Oceanography	7:30 p.m.	Salon A/B/C
Animal Bioacoustics	7:30 p.m.	Salon F/G/H
Architectural Acoustics	7:30 p.m.	Studio 9
Musical Acoustics	7:30 p.m.	Studio 4
Physical Acoustics	7:30 p.m.	Balcony L
Psychological and Physiological Acoustics	7:30 p.m.	Balcony M
Structural Acoustics and Vibration	8:00 p.m.	Studio 7

Committees meeting on Wednesday, 6 December

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Balcony M
Signal Processing in Acoustics	7:30 p.m.	Salon D

Committees meeting on Thursday, 7 December

Committee	Start Time	Room
Noise	7:30 p.m.	Studio 2
Speech Communication	7:30 p.m.	Salon A/B/C
Underwater Acoustics	7:30 p.m.	Salon F/G/H

Session 3aAA

Architectural Acoustics: Restaurant Acoustics

Andy Chung, Cochair
HKUST, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong

Siu Kit Lau, Cochair
Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Drive, Singapore 117566, Singapore

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

Chair's Introduction—7:45

Invited Papers

7:50

3aAA1. Analyses of crowd-sourced sound levels, logged from more than 2250 restaurants and bars in New York City. Gregory Farber (SoundPrint, PO Box 533, Lincoln, MA 01773, greg@soundprint.co) and Lily M. Wang (Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, Lincoln, NE)

Media reports in the United States and the United Kingdom have reported increasingly high sound levels in restaurants and bars over the past ten years, but accurate sound measurements are lacking. The Zagat survey found noise to be the second most common complaint among diners, barely behind poor service. This paper presents sound level measurements from more than 2250 restaurants and bars in New York City, using the novel SoundPrint smartphone app. The average sound level was found to be 78 dBA in restaurants and 81 dBA in bars. These sound levels do not allow ready conversation and pose a auditory health danger for noise-induced hearing loss and other non-auditory health issues. The reported sound levels by venue managers generally underestimated actual sound levels. Of interest are the findings that venues in certain neighborhoods and also of certain types of cuisine tend to be louder or quieter than others. The sound level values measured by the SoundPrint app have been tested against class 1 sound level meters and found to be reasonably close (within 1–2 dB). This report is a proof-of-concept study of crowd-sourced sound measurements, which can provide valuable data for the general public and health officials.

8:10

3aAA2. Study on influence factors of popular Chinese restaurants. Yude Zhou, Weichen Zhang, Zhiyue Shao, and Wenying Zhu (Shanghai Acad. of Environ. Sci. and Shanghai Eng. Res. Ctr. of Urban Environ. Noise Control, 508,Qinzhou Rd., Shanghai 200233, People's Republic of China, zhouyd@saes.sh.cn)

There are great differences in restaurant soundscape between China and the United States, as Chinese people prefer to be engaged in an animated conversation during dining. This study emphasizes on revealing the relationship between restaurant soundscape and local culture, dining custom and restaurant arrangement, etc., and identifying the unique characteristic of the restaurant soundscape in China. Typical Chinese restaurants are selected to investigate the dining culture in China, and to further analyze the relationship between Chinese restaurant soundscape and people's opinion toward it by both objective and subjective soundscape parametric study. It is shown that the high sound pressure lever, for instance, 70–80 dBA in dining lobby with capacity crowd, in Chinese restaurant is mainly due to a long reverberation time. Meanwhile, since dining in restaurant is regarded as a popular social activity, and people like to gather in a restaurant and chat merrily when dining, the larger table people sit by, the louder voice people have to speak in. It is also found that for the most of the time, people enjoy in the conversation, not bothered by the high sound pressure level, and if a silent dining environment is required, a private room is often available. Therefore, there is a close connection between the local culture and restaurant soundscape in China, of which better understanding can be obtained only if you are involved in.

8:30

3aAA3. Case studies that explore the soundscape of dining. Keely Siebein and Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com)

The acoustics of restaurants is related to the soundscape of dining. The goal of this study is to better understand and document the soundscape of dining as a method to analyze communication paths among diners. This will be used to determine possible acoustical interventions based on the perceived acoustical qualities of the spaces by the local experts, i.e., diners, staff, and operators of the facilities. This study compares case studies of three restaurants using the soundscape method to explore links between qualitative evaluations of the soundscape by users, staff, and operators of the facilities and acoustical metrics including analysis of impulse response measurements between occupants in the rooms, and acoustical measurements calculated from the impulse responses including STI and RT. Using the soundscape methods, along with impulse response measurements based on actual communication paths, allows for diagnosis of surfaces to be treated and allows acoustical metrics to be derived that reflect the qualitative analysis of the space as it is heard by diners, staff, and owners of the restaurants.

8:50

3aAA4. Describing sonic environment of Chinese restaurants using a mathematical model. Andy Chung (Smart City Maker, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong, ac@smartcitymaker.com) and W. M. To (Macao Polytechnic Inst., Macao, Macao, Macao)

A restaurant is commonly chosen as a venue for gathering because it serves both the dining and meeting purposes in one go, and is good for people who are busy and keen on balancing social, work, and family lives with limited time. Depending on the anticipating nature of meeting, people tend to choose a restaurant with the right mood for communication as well as the general perception of the sonic environment in whether it favors quiet conversations or louder chats. This paper presents the results of a survey on the sonic environment of a number of Chinese restaurants in Hong Kong and a mathematical model for the associated description.

9:10

3aAA5. Restaurants, bars, and dives—Can you hear me now?? Kenneth P. Roy (Armstrong, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrongceilings.com)

The acoustics of "places of public accommodation" including restaurants, bars, and dives have been an ongoing issue with the users of such places for many years. ASTM International once considered writing a standard for the measurement/performance of such spaces, but it died an uncertain death during development. More recently, the ASA was requested to look at the possibility of writing such a standard, and to that end a special session on "restaurant acoustics" sponsored by the ASA Panel on Public Policy was conducted in Boston. So what to do?? More data are obviously needed—but to what end?? It may be possible to write a measurement standard, probably at ASTM, and it may be possible to write a performance standard, probably at ASA. But, this would be a tall task given the variety of such places and the diffused focus between entertainment and "hunger and thirst" … maybe a rating system would be a more appropriate starting point. A proposal for that approach will be made herewith.

9:30

3aAA6. Speech intelligibility in restaurants—A reverberation or early reflection problem? Peter Mapp (PMA, 101 London Rd., Copford, Colchester co61lg, United Kingdom, petermapp@btinternet.com)

Several acoustic factors coincide to create hostile sonic environments in restaurants—including noise & reverberation. Reverberation time is often cited as the cause of noise build up and resultant lack of speech ineligibility in restaurants and similar spaces. This paper suggests that in smaller dining rooms and spaces for socializing, it is the generation of strong early reflections and occupation density rather than reverberation time that are the key factors. Ceiling height is also shown to be a significant factor. The paper presents an acoustic analysis of three dining/socializing spaces and the effect that the application of acoustic treatment has on the ambient noise levels and potential intelligibility. A rough rule of thumb for the volume per person required for satisfactory speech conditions is also presented together with case history data.

9:50–10:05 Break

10:05

3aAA7. Speech recognition in reverberation and background chatter. Paul Battaglia (Architecture, Univ. at Buffalo, 31 Rose Ct Apt. 4, Snyder, NY 14226, plb@buffalo.edu)

The subjective impression of acoustical comfort, as shown by previous surveys of restaurant patrons, can be achieved within a narrow range of reverberation time. These surveys also indicated that acoustical comfort is relatively unrelated to the level of background noise. In order to more fully understand the relative effects of reverberation time and noise levels, an experiment was conducted by undergraduate architecture students for speech intelligibility, not acoustical comfort, in a test space where reverberation times and levels of background chatter could be varied. The resulting data suggest that there is a threshold level of reverberation time for speech intelligibility when signal-to-noise ratios are very low, and even negative in value. This finding has consequences for the acoustical design of restaurants where the sounds of social activity are welcomed, but where background chatter should not overwhelm the ability to discern and understand nearby speech.

10:25

3aAA8. Sounds delicious: A crossmodal perspective on restaurant atmospherics and acoustical design. Steve Keller (iV, 622 Hamilton Ave., Nashville, TN 37203, skeller@ivaudiobranding.com) and Charles Spence (Crossmodal Res. Lab., Univ. of Oxford, Oxford, Oxfordshire, United Kingdom)

Recent advances in crossmodal science have demonstrated that our perception of the world around us is an amalgam of sensory experiences. Nowhere is this more evident than in the emerging discipline of Gastrophysics: the new science of eating, which blends gastronomy (the "art of the table") with psychophysics (the relationship between physical stimuli and mental phenomena). Examining restaurant atmospherics and acoustical design through the lens of crossmodalism and gastrophysics, it soon becomes clear that, when it comes to the pleasures of the table (i.e., of eating and drinking), what we put into our ears (i.e., what we hear) can be just as important as what we put into our mouths. This presentation highlights the latest research into the ways that sound and its application in restaurants

can create expectations, shape perceptions, and influence the behavior of patrons. From sonic seasonings and sensplorations, to sound systems and acoustical treatments, we will explore how sonic environments will continue to play an increasingly important role in the dining experience.

10:45

3aAA9. A comparative study of sound environment of restaurants in Singapore, Macao, and Hong Kong. Siu Kit Lau (Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Dr., Singapore 117566, Singapore, slau@acousticsresearch.com), W M To (Macao Polytechnic Inst., Macao, Macao), and Andy Chung (Smart City Maker, Copenhagen, Denmark)

Some of the key elements affecting the indoor sonic environment of a restaurant depend on its size, room configuration, architecture features, and occupancy rate. Whether the languages of people speak in the same venue or a combination of theirs have an impact remains unsure. Surveys have been conducted in three tourist cities, namely, Singapore, Macao, and Hong Kong, to collect the audio, visual, and quantitative data of the indoor sound environment of selected restaurants. This paper presents the results and comparative analytics of the surveys.

Contributed Papers

11:05

3aAA10. Quantifying the acoustic environment for restaurant consultation. Justin T. Dubin and Eli Willard (Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, justin.dubin@utexas.edu)

This talk presents a case study in which methodology for the diagnosis and treatment of poor restaurant acoustics is discussed. The subject of this case study is Uchi, an upscale Japanese restaurant in Austin, Texas. While revered for their excellent cuisine, Uchi's reputation is marred by its excessively loud dining room at peak business hours. The discussion details how the restaurant's acoustical quality was evaluated through strategic integrated impulse response measurements, along with finite element and analytical models. The measurements and models provide strong evidence of acoustical problems such as room modes, focusing gain, and excessive reverberance. Once diagnosed, improvement criteria are defined using well-established metrics for restaurants, such as "acoustic capacity" and "preferred signal-to-noise ratios." Treatment options are assessed based on their potential effectiveness as well as their feasibility to be implemented by restaurant management.

11:20

3aAA11. Sound pressures generated by exploding eggs. Anthony Nash and Lauren von Blohn (Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, anthony.nash@cmsalter.com)

Manufacturers of microwave ovens caution people to avoid re-heating certain food products because the rapid heating process can pose a danger to the user. Examples of such products are potatoes and eggs. Heating a potato in a microwave can generate steam under pressure. The internal steam pressure induces high tensile stresses in the potato skin, sometimes leading to its sudden (and unpredictable) bursting. A re-heated hard-boiled egg can also explode unpredictably but its bursting mechanism works differently than the potato. It is now believed that the egg yolk develops many small pockets of superheated water, leading to an increasingly unstable condition. When the egg yolk is disturbed by an internal or external stimulus, the pockets spontaneously boil, thereby releasing considerable energy (i.e., an explosion). An acoustical investigation was conducted using nearly 100 eggs that were re-heated under controlled conditions in a calibrated microwave oven. About a third of the re-heated, boiled eggs exploded outside the oven. For those eggs that did explode, their peak sound pressure levels ranged from 86 up to 133 decibels at a distance of 300 mm. The paper will describe the test protocols and discuss the results.

11:35–12:00 Panel Discussion

Session 3aAB

Animal Bioacoustics: General Bioacoustics

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

Contributed Papers

8:30

3aAB1. Exposure-induced changes in laboratory mouse ultrasonic vocalizations. Kali Burke, Laurel A. Screven, and Micheal L. Dent (Psychology, Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu)

Mouse ultrasonic vocalizations (USVs) have been categorized by researchers based on their variable spectrotemporal features, including frequency, intensity, and duration. These USVs may be important for communication, but it is unclear whether the categories that researchers have developed are relevant to mice when trying to understand their USVs, or if it is other properties such as the number, rate, peak frequency, or bandwidth of the calls that are significant. The current study aims to create a comprehensive catalog of the USVs that mice are producing in order to better understand if and how these animals are using their USVs for communication. Forty male and female adult CBA/CaJ mice were recorded for five minutes following either a one-hour period of isolation or exposure with a same- or opposite-sex mouse. Vocalizations were categorized into nine categories and quantified based on the bandwidth, duration, peak frequency, and total number of calls across those categories. There were significant differences in how mice produced their vocalizations when they were alone compared to following a social encounter. Further, this study provides critical data from female mice producing calls, an often overlooked phenomenon in mouse vocalization research.

8:45

3aAB2. The whistled source of Gibbon vocalizations. Shi Yu and Didier Demolin (ILPGA, LPP sorbonne nouvelle, 19 Rue des Bernardins, Paris 75005, France, camillus.shi.yu@gmail.com)

Gibbons vocalizations are often referred to as “songs” because of their pitch modulations. The elements composing Gibbons vocalizations are often called notes. Here, we discuss how they are generated. The audio waveform of some characteristic song notes of gibbons (*espèce*) show that they are produced with a high fundamental frequency (F0), often above 500 Hz, and a set of harmonic components. The F0 of these notes can also be modulated across large intervals of pitch. Spectrograms and the shape of the audio waveform suggest that they are the outcome of a whistled source. This is confirmed by the observation of audio wave from which is not complex. This also explains why there are no formants in most of their vocalizations. The high F0 source makes that the distance between harmonics and the short vocal tract length prevents the presence of resonance modes. One way to explain this whistle source is to consider that the airflow passes through stiff nonvibrating vocal folds. This setting also suggests that pitch modulations of Gibbons songs might be produced by changing the vocal folds length during their vocalizations. The effect of this change in length is increase or decrease F0.

9:00

3aAB3. Song bird vivaria considerations. Christopher L. Barnobi (Noise and Air Quality, Dudek, 1102 R St., Sacramento, CA 95816, cbarnobi@dudek.com), Katherine Snyder, and Nicole Creanza (Biological Sci., Vanderbilt Univ., Nashville, TN)

The design of laboratory animal vivaria should include consideration of factors that may impact the results of intended experiments. Two examples where the acoustic environment of birds could significantly affect experimental results include (1) song learning accuracy in isolated juvenile songbirds and (2) the preferences for bird songs by mates. We conducted a detailed analysis of a proposed vivarium design in a laboratory setting to assess important acoustical factors. We analyzed vivaria for room acoustics and sound isolation properties. Sound isolation and reverberation were primary concerns since bird song learning accuracy was an aim of the laboratory research. The environments are planned to be used to research juvenile birds learning a single model song during development. We tested the effects of treating the vivaria with sound absorbing foam, including the reverberation in enclosures and associated impacts on results comparing recorded bird songs. In addition, we tested the existing background sound levels observed in the laboratory and the sound isolation properties of the vivaria. Sound isolation results for the initial isolated environments shows transmission loss values from the exterior to the interior of the vivaria were significantly less than the expected 30 dB. Reverberation results reveal a distinct reduction when the interior of the vivarium enclosures were treated with a sound absorber.

9:15

3aAB4. Behavioral and electrophysiological assessment of hearing in the Japanese quail (*Coturnix japonica*). Sarah N. Strawn and Evan M. Hill (Psych., Univ. of Nebraska-Kearney, 2507 11th Ave., Psych. - COPH, Kearney, NE 68849, strawnsn@lopers.unk.edu)

Presented are the behavioral audiogram and auditory brainstem response (ABR) recordings from the Japanese quail (*Coturnix japonica*). This study was particularly interested in the sensitivity of this species to infrasound (i.e., frequencies below 20 Hz). Previous histological research found that the basilar papilla of gallinaceous species tends to be specifically tuned to frequencies below 1 kHz. Consistent with this finding, previous behavioral tests of chickens demonstrated exceptional sensitivity to low-frequency sounds, especially infrasound. The results of this study show that the Japanese quail has a sensitivity to 16 Hz that is equivalent to that of humans, and thus their infrasound sensitivity is far less than that of the chicken. The ABR recordings did not match the sensitivity or frequency range of the behavioral audiogram, though they provided a reasonable estimate of the high-frequency limit of the animal’s hearing range and frequency of best sensitivity. These findings are consistent with histological evaluations of the inner-ear physiology of gallinaceous birds that suggest the low-frequency end of the Japanese quail’s hearing range is shifted slightly away from that of other birds of this order, though they would still possess good sensitivity below 1 kHz.

9:30

3aAB5. An information-theoretic approach to choosing spectrogram resolutions for analyzing different biological and musical sounds. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

The great diversity and abundance of animal sounds have been portrayed by spectrograms for more than 75 years. Until recently, the time- and frequency-resolution of spectrograms was dictated by hardware limitations such as the number of filter banks in a spectrograph, the sample rate of a digitizer, or the storage capacity of a computer's memory. Improvements in these technologies, especially in the sample rate, has expanded the range of practical time- and frequency-resolutions that spectrograms can display, even if it is still necessarily caught in a tradeoff between resolving temporal or frequency details at each given sample rate. The region of this tradeoff, however, may now actually exceed the time- or frequency-resolving ability of the auditory systems of the animals that emit and perceive biological sounds. Animals are also bound by the fundamental time versus frequency tradeoff. Therefore, if we can measure the best time- and frequency-resolution to analyze a signal, we can both increase analytical efficiency and gain insight into the perceptual abilities of the receivers of the signal. An information metric is proposed for determining the optimal time- and frequency-resolution to analyze a sound. It consists of the frequency excursion of the sound across time windows, divided by the autocorrelation among adjacent spectrogram time windows. The metric is evaluated by applying it to several species of bird song, as well as whale calls, human speech, and musical notes,

9:45

3aAB6. Simulated anthropogenic noise exposure to marine invertebrates using a standing wave tube. Georges Dossot (NUWC, 1176 Howell St., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), Jason Krumholz (McLaughlin Res. Corp., Newport, RI), David Hudson (The Maritime Aquarium at Norwalk, Norwalk, CT), and Darby Pochtar (Univ. of Rhode Island, Kingston, RI)

The experimental design of a standing wave tube suitable for monitoring the impact of anthropogenic noise upon marine invertebrates is presented. Human usage of coastal water bodies continues to increase and many commercially harvested invertebrates face a broad suite of anthropogenic stressors (e.g., warming, pollution, acidification, and fishing pressure). Underwater noise is one such stressor that exists in coastal areas, but the potential impact on invertebrates, including sublethal effects such as masking, behavioral, and physiological impacts, is not well understood. A major obstacle to further progress in this field is that *in-situ* experiments using high sound levels require extensive permitting and can be difficult to monitor, while ex-situ laboratory experiments do not often account for acoustic artifacts likely present in closed tank environments. We demonstrate the design and implementation of a relatively inexpensive standing wave tube approach, which creates a uniform sound field large enough to allow simultaneous exposure of multiple invertebrates per trial. We exposed juvenile and sub-adult blue crabs (*Callinectes sapidus*) and American lobsters (*Homarus americanus*) to simulated low-frequency boat noise and mid-frequency sonar, and measured behavioral and physiological responses, as well as acoustic pressure and particle motion to fully quantify the impacts of the sound field.

10:00

3aAB7. Examination of behavioral response of wild river herring to sonar signals using acoustic telemetry. Joseph Iafrate, Stephanie L. Watwood (Navy, 1176 Howell St., Newport, RI 02717, joseph.iafrate@navy.mil), Jessica Kutcher (McLaughlin Res. Corp., Middletown, RI), and Georges Dossot (Navy, Newport, RI)

The potential effect of high-intensity noise and disturbance to fish populations is of growing concern. Adult river herring were exposed to mid-frequency (1–10 kHz) sonar signals to assess behavioral response in their natural environment. In this case study, acoustic telemetry was employed to measure fine scale movement of free-ranging river herring released in Dodge Pond, Connecticut, in response to a controlled sound exposure experiment. Alewife (*Alosa pseudoharengus)* was selected as the target species due to natural occurrence in the study area and documented hearing specializations in this species. An acoustic telemetry array was used to examine movement, spatial distribution, and schooling behavior of the fish before, during, and after exposure to mid-frequency sonar or similar sonar signals. Movement parameters examined include directional response, distance moved, swim speed, and distribution of the school. The sound field was mapped to assess received levels including both sound pressure level (SPL) and particle velocity within full extent of the telemetry array. Preliminary results will be presented including novel methods for gastric tagging, baseline behavior, performance of the array, and measurements supporting sound field mapping.

10:15

3aAB8. Opportunistic underwater recording of what might be a distress call of *Chelonya mydas agassizii*. Amaury Cordero Tapia and Eduardo Vivas (CIBNOR, Av Instituto Politecnico Nacional 195, Playa Palo de Santa Rita Sur, La Paz, Baja California Sur 23096, Mexico, evivas@cibnor.mx)

All around the world, sea turtles are considered endangered species since their population has declined in the last two decades. In Baja California Sur Mexico, there is a conservation program run by Government Authorities, Industry, and Non-Governmental Agencies focused on vulnerable, threatened, and endangered marine species. In zones of high density of sea turtles, special nets, which allow them to surface for breathing, are deployed monthly for monitoring purposes. Nets are checked every 2 hours during the 24 hours of the census. During one of these checks, a female specimen of *Chelonya mydas agassizii* was video recorded using an action cam. Posterior analysis of the recording showed a clear pattern of pulsed sound when the diver was at close proximity to the turtle. The signal covers a frequency range of 300 to 800 Hz, which is within the ABRs average audiogram range reported by Ketten and Bartol for *Chelonya mydas* and within the behavioral Audiogram for *Chelonya mydas agassizii* that we have previously reported. The video and the sound analyses of this opportunistic recording, which might be a distress call, are presented.

Session 3aAO

Acoustical Oceanography: Munk Award Lecture

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

Chair's Introduction—11:00

Invited Paper

11:05

3aAO1. New platforms, technologies, and approaches for remote inference of physical and biological parameters using acoustic scattering techniques. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

Active narrowband acoustic scattering techniques have been used for decades to study the distribution of marine organisms, such as fish and zooplankton, and to image physical oceanographic processes, such as internal waves and microstructure. In the last decade or so, these techniques have been extended to the use broadband acoustic scattering techniques for more accurate inference of relevant biological and physical parameters, such as size or abundance of organisms or intensity of mixing. Rapid advances in instrumentation and deployment platforms have also enabled new insights to be gained. In this presentation, a brief overview of this research area is given. Then, results from a decade of work on the development and implementation of broadband scattering techniques for studying physical and biological processes over relevant spatial and temporal scales are presented. Finally, recent data from an estuarine plume, collected with a broadband sonar integrated onto a Remus-100 Autonomous Underwater Vehicle, are presented. Advances and limitations of new platforms and sensors to the future of acoustical oceanography are considered.

Session 3aBA

Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications III

Pierre Belanger, Cochair

Mechanical Engineering, Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Guillaume Haiat, Cochair

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

Chair's Introduction—8:25

Invited Papers

8:30

3aBA1. Numerical simulation of the ultrasonic propagation in bone tissue. 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 bone has a complex structure, it is difficult to model analytically the behavior of ultrasound propagating in bone although ultrasound is useful for the diagnosis of not only bone density but also bone quality. Our group, therefore, had been working on simulating ultrasound propagation inside cancellous bones and models with cortical bones using actual mammal bones. In this paper, the basis and the results of the 3-D elastic FDTD (finite-difference time domain) method will be presented. The FDTD simulation only requires the 3-D geometry of the model and the distribution of acoustic parameters (density, speed of longitudinal wave, and shear wave) of the media such as bone and bone marrow, so that the effect of the each acoustic parameter or the geometry (e.g., BV/TV) can be easily investigated by changing these values. In addition, the effect of the frequency-dependent absorption caused by the viscosity and the piezoelectricity of bone also can be considered in the viscoelastic FDTD and the piezoelastic FDTD, respectively. The effects of these characteristics are not negligible. Moreover, thanks to the recent progress of the PC resource, a real-size simulation of human radius model is realized. Some preliminary data will also be presented. [JSPS KAKENHI 16K01431.]

8:50

3aBA2. Two ultrasound longitudinal waves in cancellous bone acquired using a fast decomposition method with a phase rotation parameter for bone quality assessment. Hirofumi Taki (Biomedical Eng. for Health and Welfare, Graduate School of Biomedical Eng., Tohoku Univ., 2-1 Seiryo-machi, Aoba-ku, Sendai, Miyagi 980-8575, Japan, hirofumi.taki.a1@tohoku.ac.jp), Yoshiki Nagatani (Dept. of Electronics, Kobe City College of Technol., Kobe, Hyogo, Japan), Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Shin-Ichi Izumi (Biomedical Eng. for Health and Welfare, Graduate School of Biomedical Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Ultrasound signal passing through cancellous bone consists of two longitudinal waves: fast and slow waves. Accurate decomposition of the fast and slow waves is supposed to be highly beneficial in order to determine the characteristics of cancellous bone. We applied a fast decomposition method using adaptive beamforming technique with a phase rotation parameter to ultrasound signals that passed through bone specimens with various bone volume to total volume (BV/TV) ratios in a simulation study. The decomposition method accurately characterized the two waves with the normalized residual intensity of less than −19.5 dB when the specimen thickness ranged from 4 to 7 mm and the BV/TV ratio was from 0.144 to 0.226. The ratio of the peak envelope amplitude of the fast wave to that of slow wave increased monotonically as BV/TV ratio increased. The result also indicates a strong relationship between the phase rotation value and BV/TV ratio, where the variation of the phase rotation value increased as specimen thickness increased. These findings show that the decomposition method using adaptive beamforming technique with a phase rotation parameter has the high potential in estimating the BV/TV ratio in cancellous bone.

9:10

3aBA3. Bone repair and ultrasound stimulation: An insight into the interaction of LIPUS with the bone callus through a multiscale computational study. Cécile Baron (ISM AMU-CNRS, 163 Ave. de Luminy, Marseille 13288, France, cecile.baron@univ-amu.fr), Carine Guivier-Curien (IRPHE CNRS-AMU, Marseille, France), Vu-Hieu Nguyen, and Salah Naili (MSME CNRS-UPEC, Créteil, France)

In the 1950s, the effect of ultrasound stimulation on bone healing has been discovered. Nowadays, Low Intensity Pulsed Ultrasound Stimulation (LIPUS) is admitted to influence the mechanotransduction of bone. Nevertheless, despite a growing literature—cell cultures, animal models, and clinical studies—the underlying physical and biological mechanisms of LIPUS on bone healing are still misunderstood. Inspired from previous studies on the mechanotransduction induced by physiological loading, this work focuses on the effect of LIPUS on the osteocytes. These bone cells are thought to be the principal mechanosensors of bone. They are ubiquitous inside the bone matrix, immersed in the lacuno-canalicular network (LCN) filled with interstitial fluid (IF). The goal is to relate the ultrasound stimulation applied at the tissue scale, to the biological response at the cell scale. To tackle this question, two finite element models were implemented in the commercial software Comsol Multiphysics. The tissue-scale model considers an anisotropic poroelastic matrix to evaluate the IF pressure gradient induced by LIPUS into the LCN. Then, in the cell-scale model, the IF shear stress magnitude and the induced drag forces applied on osteocyte process are calculated and compared with levels of cell activation recorded in literature.

Contributed Paper

9:30

3aBA4. Semi-analytical finite-element based method for inverse characterization of cortical bone using low-frequency guided waves. Daniel Pereira (Dept. of Mech. Eng., École de technologie supérieure, 100 Rue Notre-Dame O, Montreal, QC H3C 1K3, Canada, pereira.ufrgs@gmail.com), Julio Fernandes (Dept. of Surgery, Univ. of Montreal, Montreal, QC, Canada), and Pierre Belanger (Dept. of Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

Axial transmission research has demonstrated that low-frequency ultrasonic guided waves are sensitive to changes in the intracortical bone, which is of interest since the resorption in the endosteal region is associated to early-stage osteoporosis. Current methods rely on inversion schemes used to match experimental data with the theoretical data obtained from simplified models. However, due to the importance of the cross-sectional curvature of the cortical bone at low-frequency (e.g., $<$200 kHz), the implementation of a more elaborate model remains an open issue. Thus, the aim of this paper is to introduce a semi-analytical finite-element (SAFE) model to be used along with a genetic algorithm for the inverse characterization of cortical bone. Our proposal is to validate an inverse scheme using laboratory-controlled measurements on bone-mimicking phantoms at low frequency. An arbitrary cross-sectional geometry, instead of a plate or cylinder simplification, was implemented. Despite a computationally expensive SAFE routine, the results show that the model outputs estimated by the genetic algorithm are in good agreement with the reference values obtained by μCT images. The possibility of implementing parallel computation using graphics processing units in order to increase the level of complexity of the SAFE model may now be investigated.

Invited Papers

9:45

3aBA5. Ultrasonic bandgaps and interlaminar interface echoes of composite laminates: Analysis and experiments. Shiro Biwa (Dept. of Aeronautics and Astronautics, Kyoto Univ., C-Cluster III, Katsura, Nishikyo-ku, Kyoto 615-8540, Japan, biwa@kuaero.kyoto-u.ac.jp) and Yosuke Ishii (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi, Japan)

Increasing applications of carbon fiber reinforced plastics (CFRP) in aerospace industry highlight the importance of nondestructive methods to characterize their mechanical properties as well as defect concentrations. In CFRP laminate structures, thin resin layers, typically of a few micron thickness, are present between plies. Their spatially periodic nature causes bandgaps for ultrasonic waves having wavelengths comparable to the ply thickness. Features of the reflection and transmission spectra of ultrasonic waves including such bandgaps can be utilized for nondestructive characterization of CFRP laminates. In particular, the oscillatory frequency dependence of the reflection or transmission spectra of ultrasonic waves near the band gaps can be used to identify the equivalent stiffnesses of interlaminar interfaces. These interfacial stiffnesses contain rich information of the mechanical soundness of the interlaminar interfaces which are otherwise difficult to evaluate with traditional ultrasonic methods working in relatively low frequencies. The evidence of ultrasonic bandgaps of CFRP laminates can also be observed as long-standing signals following the surface echo in the temporal reflection waveforms of CFRP laminates (interlaminar interface echoes). In this presentation, the physical phenomena behind the ultrasonic bandgaps and interlaminar interface echoes of CFRP laminates are discussed together with their applications to nondestructive characterization of these materials.

10:05–10:25 Break

10:25

3aBA6. Remote sparse distributed sensors to image bondline defects between a composite panel and a stiffener. Sreedhar Puliyakote (I2M, Univ. of Bordeaux, 351 cours Liberation, I2M (A4) - Univ. of Bordeaux, Talence 33400, France), Xudong Yu, Zheng Fan (Nanyang Technolog. Univ., Singapore, Singapore), and Michel Castaings (I2M, Univ. of Bordeaux, Talence, France, michel.castaings@u-bordeaux.fr)

Adhesive bonding is widely used to fix skins to reinforcing elements, like stiffeners, in aerospace composite structures. A well-cured bond offers uniform stresses, good joint strength, and improved fatigue and impact resistance, and is therefore crucial to the performance of the entire structure. In the previous work by the authors, ultrasonic feature guided wave (FGW) has been discussed as a screening tool for quick inspection of the bondline between a CFRP panel and stiffener. However, it was found that material damping and weak reflections by defects along the feature makes the use of a single transducer in pulse echo mode quite inefficient. In this study, a structural health monitoring (SHM) technique is proposed to inspect the bondline, based on the radiation of plate modes into the CFRP panel when an incident FGW interacts with a defect in that bondline. A sparse array of sensors organized on the CFRP panel away from the stiffener was used to capture the waves radiating into the panel. A synthetic focusing method was applied to process the recorded signals for imaging the damaged area in the bond. Adhesive defects with varying dimensions were successfully identified in 3D FE simulations, with supporting experimental results to validate this method.

10:45

3aBA7. Simulation of wave propagation in polycrystalline materials. Gaofeng Sha (Mater. Sci. and Eng., Ohio State Univ., Columbus, OH), Anton Van Pamel, Ming Huang (Mech. Eng., Imperial College London, South Kensington, London SW7 2AZ, United Kingdom), Stanislav I. Rokhlin (Mater. Sci. and Eng., Ohio State Univ., Columbus, OH), and Michael J. Lowe (Mech. Eng., Imperial College London, South Kensington, London, United Kingdom, m.lowe@imperial.ac.uk)

The propagation and scattering of elastic waves within heterogeneous materials is of wide interest in seismology, medical ultrasound, and non-destructive evaluation. Accurate models of the behavior of the waves are important for the development of methods to characterize the materials using this information, as well as for the development of methods pursuing other information about the materials for which the attenuation and scattering are a nuisance to be minimized. Until recently, the leading models have been limited to analytical methods based on low-order scattering assumptions. However, it has now become possible to perform accurate three-dimensional Finite Element simulations, using spatial representation at grain scale, in significant sample volumes of large numbers of grains. Recent work by the authors has demonstrated that this approach can deliver attenuation and wave speed predictions that are good enough to enable evaluations of the features and approximations of the analytical models. Current work is addressing the use of this approach, in combination with analytical models, to investigate the physical phenomena of the scattering attenuation and wave speed dispersion, including the influences of materials symmetries and degree of anisotropy, and the effects of metal forming such as grain elongation.

11:05

3aBA8. Ultrasound measurement of texture in bulk polycrystalline materials. Bo Lan, Michael J. Lowe (Mech. Eng., Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk), T. B. Britton, and Fionn P. Dunne (Materials, Imperial College London, London, United Kingdom)

Manufacturing of metal components often results in significant texture, that is, to say, preferred orientations of their polycrystals. Since each crystal can be strongly anisotropic, this can give the component orientation-dependent material properties, affecting stiffness, thermal expansion, strength, and fatigue and creep resistance. So, it is important to be able to measure texture, especially for high-value safety-critical components. Typically, the Orientation Distribution Function (ODF) of the crystals can be measured on exposed surfaces using EBSD, or in thin samples using neutron diffraction. But both are expensive, and until recently there has not been a means to measure ODFs internally in bulk materials. However, the authors have recently developed a method to determine internal texture from measurements of wave speeds at selected angles through the volume of the material. This is based on a convolution of wave speeds in a single crystal with the ODF, giving the resultant polycrystal wave speed angular function. The principles of the method have been established and validated for single phase hexagonal and cubic materials. Ongoing work is investigating the use of multiple wave modes (shear and compressional) to identify further information; the extraction of volume fraction and texture of two-phase materials; and the pursuit of further refinement of the texture results by improved knowledge of the single crystal properties.

Contributed Paper

11:25

3aBA9. Impact of sol-gel transition on the acoustic properties of complex model foods: Application to agar/gelatin gels and emulsion filled gels. Mathieu Mantelet, Maud Panouillé, François Boué (UMR 782 GMPA, INRA - AgroParisTech - Université Paris Saclay, UMR 782 GMPA, 1 Ave. Lucien Brétignières, Thiverval-Grignon 78850, France, mathieu.mantelet@inra.fr), Frédéric Restagno (UMR 8502 LPS, CNRS - Université Paris Sud - Université Paris Saclay, Orsay, France), Isabelle Souchon, and Vincent Mathieu (UMR 782 GMPA, INRA - AgroParisTech - Université Paris Saclay, Thiverval-Grignon, France)

Quantitative Ultrasound techniques are good candidates for the *in situ* and real-time mechanical characterization of tongue-food-palate system, and thus to improve the understanding of the determinants of texture perception of food. Different model foods (consisting in gels and emulsion filled gels composed of agar and/or gelatin) have been designed for their contrasting properties in terms of texture perceptions. Prior to the feasibility study of a Quantitative Ultrasound method to monitor their mechanical breakdown during a compression, the aim of this study is to determine the respective roles of structure and mechanical properties of the different model foods in the variations of ultrasonic wave properties. Ultrasonic velocity, reflectivity, and attenuation were monitored during the sol-gel transition (from 50°C to 20°C) at 1 MHz in pulse-echo mode, and were confronted to visco-elastic moduli and mass density measurements. The results put in evidence the role of biopolymer concentration (independently from Young's and shear moduli) on the variations of velocity and reflectivity, resulting from joint variations of mass density and bulk modulus. Moreover, the ultrasonic attenuation was confirmed to depend on molecular relaxation phenomena of water, which are important in high-concentration gelatin samples.

Session 3aED

Education in Acoustics: Hands-On Acoustics Demonstrations for Middle- and High-School Students

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session, "Hands-On" demonstrations will be set-up for a group of middle- and high-school students from the New Orleans area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should e-mail Keeta Jones (kjones@acousticalsociety.org).

WEDNESDAY MORNING, 6 DECEMBER 2017 STUDIO 7, 7:40 A.M. TO 12:00 NOON

Session 3aID

Interdisciplinary and ASA Committee on Standards: Standards: Practical Applications in Acoustics

Christopher J. Struck, Cochair
CJS Labs, 57 States St., San Francisco, CA 94114

Robert D. Hellweg, Cochair
Hellweg Acoustics, 13 Pine Tree Road, Wellesley, MA 02482

Chair's Introduction—7:40

Invited Papers

7:45

3aID1. A history of noise exposure standards. William J. Murphy (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), Amanda Azman (Pittsburgh Mining Res. Div., National Inst. for Occupational Safety and Health, Pittsburgh, PA), and Mark R. Stephenson (Stephenson and Stephenson Res. and Consulting, LLC, Loveland, OH)

In the 1950s, the Committee on Hearing, Bioacoustics and Biomechanics (CHABA) investigated and developed recommendations for a variety of noise exposures. In 1956, the United States Air Force published AF Regulation 160-3, generally considered to be the first hearing conservation program in the US. The Walsh-Healy Act regulations [41 CFR 50-204.10] defined noise limits for occupational noise exposure for government supply contracts and the Federal Coal Mine Health and Safety Act of 1969 (Public Law 91-173) adopted these limits for underground and surface coal mine operations. The Occupational Safety and Health Act of 1970 (Public Law 91-596) created the Occupational Safety and Health Administration (OSHA) and the National Institute for Occupational Safety and Health (NIOSH). NIOSH published *Criteria for a Recommended Standard: Occupational Exposure to Noise* in 1972. OSHA promulgated a regulation for occupational noise exposure and updated that with the Hearing Conservation Amendment 29 CFR Part 1910.95. The Mine Safety Administration has regulations for noise exposure that were updated in 1999, 30 CFR part 62. The military developed the materiel acquisition standard MIL-STD 1474E for high-level impulse noise. Existing and future needs for noise exposure standards for continuous, intermittent, and impulsive noise exposures will be discussed in this paper.

8:10

3aID2. An overview of ANSI/ASA S3.7-2016: Method for measurement and calibration of earphones. Christopher J. Struck (CJS Labs, 57 States St., San Francisco, CA 94114, cjs@cjs-labs.com)

An overview of the recent revision of "ANSI/ASA S3.7-2016 Method for Measurement and Calibration of Earphones" is given. Guidance for the selection of the appropriate coupler or ear simulator for a given earphone and application is provided. The various clauses within the standard describing measurements of calibrated frequency response, input-output linearity, electrical impedance, and non-linear distortion are presented. The method in the standard for dealing with issues related to positioning and test repeatability when using a head and torso simulator is described. The different standard coupling configurations for insert earphones and receivers are detailed. Intermodulation and difference frequency distortion test methods and their use to overcome the bandwidth limitations of the measurement system are also described. A method for measuring and quantifying left-right balance in a stereo headphone is also presented.

8:35

3aID3. Anechoic and hemi-anechoic chamber performance qualification—Review of recent standards changes. Douglas Winker (ETS-Lindgren, Inc., 1301 Arrow Point Dr., Cedar Park, TX 78613, douglas.winker@ets-lindgren.com)

Historically, anechoic and hemi-anechoic chamber qualification has been governed by ISO 3745 Annex A. In 2017, a revised version of ISO 26101 was published. A simultaneous update to ISO 3745 Annex A, now refers to the method described in ISO 26101. Many differences exist between the previous versions of ISO 3745 Annex A and the new ISO 26101 method including curve fit, traverse direction and quantity, traverse length, measurement spatial resolution, and others. These changes will impact qualification distances and frequency ranges for anechoic and hemi-anechoic chambers. A comparison between the previous method and the new method will be presented. The impacts of those changes on existing chambers and new chamber designs will be discussed.

9:00

3aID4. Recent developments in sound power level measurement standards. Robert D. Hellweg (Hellweg Acoust., 13 Pine Tree Rd., Wellesley, MA 02482, hellweg@hellwegacoustics.com)

The Acoustical Society of America ANSI accredited standards committee (S12) and the ISO committee TC43 SC1 (Noise) have developed a series of standards for measuring sound power levels of products. The engineering method to determine sound power levels in a free field over a reflecting plane is ANSI/ASA S12.54, which is the national adoption of ISO 3744. This is the standard "of choice" for a large majority of manufacturers of equipment—from consumer products to industrial machinery. In order to address issues found by using the standard over the years, some of which rarely occur, ANSI/ASA S12.54 (ISO 3744) has become larger and more complex. The ISO working group began a project to simplify ISO 3744 in order to make it simpler, easier to use, and more understandable. Instead of attempting to cover 99% of the situations that could arise during testing, the main body of the revised standard will address 90% of the cases. The issues that occur infrequently will be moved into annexes, and some parts will be consolidated into other standards. The current status of these changes to ISO 3744 and plans for the other sound power level standards will be discussed.

9:25

3aID5. Introduction to S12.70 speech privacy in healthcare. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com) and Eric L. Reuter (Reuter Assoc., Portsmouth, NH)

The Privacy Rule within HIPAA requires that personal identifiable information be kept private. Reasonable safeguards must be taken for electronic, paper, and oral (speech) communications. The S12.70 standard provides a means to design, operate, and enforce to this requirement. We will explore the background and practical application of this standard.

9:50

3aID6. Acoustical measurements with mobile devices and the challenge of standards compliance. Benjamin Faber (Faber Acoust., LLC, 277 S 2035 W, Lehi, UT 84043, ben@faberacoustical.com)

With more attention being brought to the use of mobile devices for acoustical measurements, questions of measurement quality and standards compliance continue to be raised. What kinds of acoustical measurements are realistic with a mobile device? Can a smartphone or tablet be used as a properly qualified sound level meter? If not, under what conditions might it be appropriate to rely on a mobile measurement solution? For the purposes of this presentation, the IEC 61672 standard for sound level meters will be discussed with respect to mobile measurement apps and devices.

10:15–10:30 Break

10:30

3aID7. Measurement uncertainty and its application to standards in acoustics. Christopher J. Struck (CJS Labs, 57 States St., San Francisco, CA 94114, cjs@cjs-labs.com)

The basic theory of measurement uncertainty as found in IEC and ISO standards is reviewed. Random error, bias error, confidence interval, coverage factor, and expanded uncertainty are defined. The concept of acceptance intervals applied to tolerance limits on the performance of electroacoustical systems for outgoing quality control and incoming inspection is presented. Calculation of the

maximum permitted uncertainty is detailed and its application in the development of an uncertainty budget for a given electroacoustical measurement is shown. Example uncertainty budgets are developed for response measurements on a number of common electroacoustical devices including loudspeakers, microphones, telephones, earphones, and hearing aids. In addition to using this method to develop an uncertainty budget, it is shown how it can also be used as a diagnostic tool to find problems in the specification of various components in a measurement chain. An example of its diagnostic use on the current version of the ANSI S3.22 hearing aid standard is shown. Last, the uncertainty is calculated for a measurement of random noise and its dependence on bandwidth and averaging time is examined.

10:55

3aID8. Upcoming international standards in psychoacoustics. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Besides loudness, other psychoacoustic parameters like tonality and roughness can be used for product noise assessments. Tonality measurement procedures quantify the audibility of prominent tonal components and roughness evaluates modulation characteristics. In June 2017, two new ISO standards for loudness were published: ISO 532-1 (Zwicker method, based on DIN 45631/A1:2010-03) for stationary and time-varying sounds and ISO 532-2 (Moore/Glasberg method, based on ANSI S3.4-2007) for stationary sounds only. Additionally, ISO TC 43/WG 9 started now to work on ISO 532-3 for time-varying loudness based on the TVL model of Moore/Glasberg. For many years, tonality measurement procedures such as the Tone-to-Noise Ratio (TNR) and Prominence Ratio (PR) have been applied to identify prominent discrete tones. Recently, a new perceptually accurate tonality assessment method based on a hearing model of Sottek was developed which evaluates the nonlinear and time-dependent loudness of both tonal and broadband components, separating them via the autocorrelation function. This new perception-model-based procedure, suitable for identifying and ranking tonalities from any sources, is proposed for the next edition of ECMA-74 as an alternative to TNR and PR. Furthermore, it is planned to extend ECMA-74 by a roughness calculation procedure based on the same hearing model approach together with some post processing (such as a weighted modulation spectral analysis). The paper gives an overview of recent developments of psychoacoustic standards.

11:20

3aID9. Development of an S3/SC1 standard for auditory evoked potential hearing tests in toothed whales. Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org)

The National Marine Fisheries service regulates the impact of anthropogenic ocean noise to marine mammals. Other government and commercial groups that produce ocean noise are subject to regulation. Both groups desire greater knowledge of hearing in marine mammals to better predict, mitigate, and regulate noise exposure. Auditory evoked potential (AEP) hearing tests have become a primary method by which hearing tests are performed in toothed whales (e.g., dolphins, porpoises). The synthesis of AEP data to address noise impacts to marine mammals is limited, in part, because of the variability between AEP thresholds measured by different researchers and laboratories using different methodologies. Methodological differences exist for AEP detection, threshold estimation, calibration, and stimulus delivery. Threshold variability could be reduced through standardization of AEP hearing test methods, which is desired by regulatory agencies and sound producers. An S3/SC1 standard for AEP hearing test methods in toothed whales has been initiated to address this need. The draft standard will be submitted to ANSI for consideration by the end of 2017. Adoption of the standard internationally would further improve threshold comparability by ensuring researchers globally utilize the same methods. The process should increase AEP data reliability for purposes of addressing ocean noise issues.

Contributed Paper

11:45

3aID10. A primary method for the complex calibration of a hydrophone from 1 Hz to 2 kHz. William H. Slater, Steven E. Crocker (Naval Undersea Warfare Ctr. Div. Newport, 1176 Howell St., Newport, RI 02841, william.h.slater@navy.mil), and Steven R. Baker (Naval Postgrad. School, Marina, CA)

Primary calibrations of hydrophones at frequencies less than about 1 kHz are typically performed in a coupler reciprocity chamber (“coupler“); a closed test chamber where time harmonic oscillations in pressure can be achieved and the reciprocity conditions required for a primary calibration can be realized. The closed and controlled environment in the coupler allows for the performance of primary calibrations over the temperature and hydrostatic pressure range found in the ocean. The coupler reciprocity system employed by the United States, in service since the 1960s, provides only the magnitude of the pressure sensitivity and not the phase. Recent work has demonstrated a method for the primary calibration of both the magnitude and phase of the complex sensitivity for a hydrophone at frequencies ranging from 1 Hz to 2 kHz. The combined expanded uncertainties of the magnitude and phase of the complex sensitivity at 1 Hz were 0.1 dB re 1V/μPa and $\pm 1°$, respectively.

Session 3aMU

Musical Acoustics: General Topics in Musical Acoustics

Whitney L. Coyle, Chair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Contributed Papers

8:45

3aMU1. Signal analysis of New Orleans jazz clarinet sounds. Joshua Veillon, Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jveillon@uno.edu), and Michael White (Xavier Univ., New Orleans, LA)

Many have noted the remarkable variety in clarinet tones produced by one player as compared to another, for example, in New Orleans jazz performances. To understand and interpret these differences, they are often described qualitatively in musical terms. These descriptions have potentially loose interpretations from one musician to another. The research presented here is an attempt to quantify such differences using acoustical signal processing techniques including Fourier transform analysis, power spectral analysis, spectrograms or time-frequency plots, and energy dependence on frequency. Variables such as recording equipment, instrument design, and performance techniques of individual artists will be considered. Examples from historical and contemporary recordings will be discussed and used to demonstrate conclusions.

9:00

3aMU2. Development process of the grand piano duplex scale. Niko Plath (Inst. of Systematic Musicology, Univ. of Hamburg, Germany, Neue Rabenstr. 13, Hamburg, Hamburg 20354, Germany, niko.plath@uni-hamburg.de) and Katharina Preller (Deutsches Museum, München, Germany)

Acoustical investigations of the earliest examples of implemented duplex schematics allow to shed light on the question of how much of the piano development in the late 19th century was still a trial and error approach or already a methodical process based on empirical findings. Further, it is investigated if the intended effect was instantly perceivable, or if it was first a theoretical concept, which was later refined to enhance a certain tonal character. For a modern grand piano, the influence of the duplex schematic on the vibroacoustic behavior and the aural impression is extensively covered by Öberg and Askenfelt. By performing measurements on several pianos from the late eighteen hundreds, most prominently the so-called Helmholtz piano, a Steinway grand presented as a gift to von Helmholtz in appreciation of the impact of his work on the company's piano development, it is possible to retrace development procedures in the workshop in that specific time period. Design parameters from the respective pianos are obtained, e.g., string length ratios, angles from main string to front duplex, boundary conditions, and string diameters. Acoustical measurements, as proposed by Öberg and Askenfelt, are performed to examine if these constructive changes have any significant effect on the generated sound and to allow a comparative analysis of both historical and contemporary instruments.

9:15

3aMU3. Acoustical nonlinearities in the structural components of the piano. Lauren Neldner, Eric Rokni, Cierra Gibson, and Thomas Moore (Dept. of Phys., Rollins College, Winter Park, FL 32789, lneldner@rollins.edu)

Anomalous frequency components in piano sound, commonly referred to as phantom partials, are generally believed to originate from a geometrical nonlinearity of the string. Recent evidence suggests that phantom partials may also be produced by nonlinearities in the non-string components of the piano such as the soundboard, case, bridge, and frame. We report experimental results indicating that in addition to phantom partials, these nonlinearities associated with the piano structure also generate harmonic frequency components, which may affect the perceived sound more significantly than the phantom partials. The possible origins of this nonlinearity will be discussed.

9:30

3aMU4. An acoustical study focused on the effects of guitar soundboard material on sound propagation. Guido Saccaggi (Mech. Eng., Catholic Univ. of America, 2000 Woods River Ln., Duluth, GA 30097, guidosaccaggi@gmail.com), Josh Capozella, Derek Kuebler, Diego Turo, and Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, Washington, District of Columbia)

This presentation describes a student project that aims to study the effect of a guitar top's material and bracing design on measurable parameters such as radiated sound power and spectrum. To illustrate, two identical, classical guitars were purchased, and the soundboards were replaced. The top of one guitar was replaced with a soundboard composed of two different woods and the other with a carbon fiber composite. Mahogany and spruce were chosen as the two woods, as mahogany is purported to enhance lower notes and spruce to enhance higher notes. Carbon fiber was picked to maximize flexural rigidity of the top. Performances of the two guitars were characterized by measuring the acoustic pressure at a fixed distance from the guitar. Additionally, the vertical acceleration of specific locations of the top was also measured as well as the strain in the pick to make a fair comparison between the sounds radiated by both guitars. Sound recordings were analyzed with time-domain, frequency spectra, and to compare the performance of the two guitars

9:45

3aMU5. Acoustic characterization of a Ukelin. Tyler J. Cottrell, Thomas Gleason, Philip P. Faraci, Eoin A. King, Robert Celmer (Acoust. Program & Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfied Ave., West Hartford, CT 06117), (faraci@hartford.edu)

An acoustical characterization was performed on a *ukelin*, a wooden hybrid stringed instrument designed to combine features of the ukulele and the violin. This instrument consists of two resonant cavities at opposite ends of the instrument, one supporting sixteen strings that are individually bowed, and the other supporting 16 strings arranged in groups of four-note diatonic chords that are simultaneously plucked with the free hand, allowing

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an individual player to produce a melody/accompaniment combination. The ukelin design was patented in the early 20th century and sold door-to door as a novelty instrument. Modal Analysis results of a sample instrument are presented and compared with an FEA normal modes solution using ANSYS. Preliminary results show good agreement of the mode shapes for several modes between the modal results and the ANSYS predictions, but a significant difference in mode frequencies for a given mode shape. The differences in frequency are attributed primarily to unknown material properties of the wood of the sample instrument. Implications of the use of a finite-element model and of the Dynamic Modification feature of modal analysis are discussed.

10:00–10:15 Break

10:15

3aMU6. Evaluating the use of crowdsourced data classification in an investigation of the steelpan drum. Joseph A. Garcia (Natural Sci., Joliet Junior College, 2707 Ruth Fitzgerald Dr., Plainfield, IL 60586, jgarcia7713@gmail.com) and Andrew C. Morrison (Natural Sci., Joliet Junior College, Joliet, IL)

The effectiveness and reliability of crowdsourced data classification to study the acoustics of the steelpan was evaluated. A project was developed and hosted on the widely used Zooniverse website. Volunteers on the project's site were asked to identify areas of maximum vibrations (called antinodes) and number of bright rings (fringes) in those areas for each classification. We explored various methods in ensuring volunteers generate successful classifications. The data for classification comes from a high-speed video recording, paired with Electronic Speckle Pattern Interferometry, of a strike on the steelpan's surface, which produces thousands of frames to be analyzed. We developed the project in preparation for a public release. We have analyzed the collected classifications using imported Python libraries. After validation and averaging of volunteer classifications, an Amplitude vs. Time graph was obtained for each antinode region, which included a comparison between using the area of the antinode region or the number of fringes as an indicator of a region's amplitude.

10:30

3aMU7. A study on the sound investigation of wooden gong rock. Bong Young Kim and Myungjin Bae (Sori Sound Eng. Lab, Soongsil Univ., 21-1, Garak-ro 23-gil, Songpa-gu, #203, Seoul 05669, South Korea, bykim8@ssu.ac.kr)

In Korea, there is a rock with a wooden gong sound. It is the wooden gong rock of Pyochungsa Temple in Miryang, Gyeongsangnam-do, and drum rock of Palgongsan Mountain in Daegu. The wooden gong is one of the Buddhist ceremonial tools that make a sound by knocking an empty tree. In this paper, we will investigate which sound component of wooden gong rock and drum rock as a wooden gong sound through sound analysis. The wooden gong resonates at about 600 Hz when tapped and continues to sound for about 0.1 second. However, most stones and rocks have a sound duration of less than 0.3 seconds without any resonance. Pyochungsa Temple's Wooden gong rock resonates at about 600 Hz and shows a duration of 0.6 to 0.9 seconds, depending on the knocking position and strength. Palgongsan Mountain's Drum rock resonates at about 400 Hz and has a duration of about 0.5 seconds. We conducted a MOS test on 30 people to see how much sound like a wooden sound for each sound. The results were as follows: wooden gong 4.5, wooden gong rock 4.1, drum rock 3.4, and common stone 2.0. Thus, the sound component of Pyochungsa Temple's wooden gong rock is closest to the wooden gong sound, so it sounds like a wooden gong sound. Wooden gong rock's wooden gong sound inspires Buddhist believers and tourists with Buddhist aspiration and patriotic loyalty. By investigating the sounds of these cultural heritages, the value and meaning of cultural heritage can be reexamined.

10:45

3aMU8. Time-scaling vibrato tones while preserving vibrato rate. Yang Shi (Elec. Eng. Dept, Univ. of California at Los Angeles, 56-125B Eng. IV Bldg., 420 Westwood Plaza, Los Angeles, CA 90095-1594, mikeshiyang@gmail.com) and James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Elongation of vibrato tones while preserving vibrato rate is a non-trivial problem in audio processing. It is well known that linear time-scaling in the time domain simply translates duration in lockstep with pitch. In the time/frequency domain, frequencies are preserved under linear time-scaling, but vibrato rates are not. Looping can preserve rate, but there is a problem synchronizing loops with vibrato cycles. Our current method is to parameterize harmonic frequency- and amplitude-vs-time variations so that vibrato rate is one of the parameters, along with vibrato depth and frequency drift. A heterodyne/filter method, where the heterodyne frequency is estimated from an FFT of the full-duration frequency deviation waveform, is used for vibrato analysis. Harmonic amplitude vibrato parameters can be estimated either by using the heterodyne/filter method directly or by using amplitude-vs-frequency relationships. Once the model is completed, the harmonic parameters can be time-scaled while keeping the vibrato rate intact. An exception to this treatment is to keep the attack and decay epochs intact. Once recomputed, the new time-varying harmonic spectrum representation is converted to a time-domain signal via sinusoidal additive synthesis.

11:00

3aMU9. A study on the song feelings change of soprano singers. Sang Bum Park and Myungjin Bae (Soongsil Univ., Sando-ro 369, Dongjak-gu, Seoul, Korea, Seoul 06978, South Korea, sbpark8510@naver.com)

The types of female vocalist are three classified into Soprano, mezzo-Soprano and Contralto according to the register. The Soprano speaks the highest voice of the female Soprano voice. In this study, we analyzed the changes in the sound width of the Coloratura Soprano, which is particularly accurate at the highest register. We measured the extent to which the world famous ten Soprano singers were attracted to the songs and presented the results to MOS SCORE. As MOS SCORE result, the more the change characteristic of the tone of the Soprano voice, the more conspicuous phenomenon becomes conspicuous. Also, the more the change in tone is displayed, the higher the extent that it is attracted by feeling the change of emotion greatly matched. By utilizing the characteristics shown in this research, the Soprano singer's prospect is inferred by referring to the Soprano singer's seamless tone change characteristics according to the correlation between tone change and fascination It became possible to do.

11:15

3aMU10. Time-scaling nonvibrato musical tones while preserving timbral texture. An Zhao (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801, anzhao2@illinois.edu) and James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Sounds produced by linear stretching of nonvibrato single tones fail to preserve their original microvariations. While looping is commonly used for elongation with wavetable synthesis, this method requires careful attention to periodic waveform boundaries. Time extension by time reversal ensures continuity in the time domain, but this method causes audible clicks due to waveform phase reversals. This problem is overcome by using a pitch-synchronous phase vocoder method that generates time-varying harmonic amplitude and frequency envelopes. Applying the time-reversal looping method to these envelopes and converting to the time domain via sinusoidal additive synthesis can generate elongated versions of the original sound without audible clicks. To retain realism, original attack and decay time data are preserved. For shortening, envelopes are cross-faded between the attack-end and decay-begin points. As an application, the algorithm has been implemented in a score-processing music synthesis program.

11:30

3aMU11. Evaluation of complex stimuli and resulting distortion product spectrum in auditory distortion product synthesis. Alex Chechile (CCRMA, Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, chechile@ccrma.stanford.edu)

Auditory distortion products are frequency components produced within the listener's ears upon the presentation of simultaneously sounding stimulus tones. Under certain conditions the listener can perceive distortion products as additional frequencies not present in the acoustic space, and the tones appear to be localized within the head. Musicians have been aware of this phenomenon of "combination tones" since the 18th century, but advancements in technology allow for new possibilities in music synthesis. *On the Sensations of Tone* is a series of compositions featuring a separate stream of musical material generated as auditory distortion products. The pieces are built using *The Ear Tone Toolbox*, a collection of open-source instruments for producing combination tones. While a single pair of stimulus frequencies typically elicits a small number of perceivable distortion products, the author puts forth a technique for evoking a spectrum of distortion products using a greater number of frequencies. The technique is informed by a mathematical model and is the basis of the next version of *The Ear Tone Toolbox*. This talk will provide an overview of the updated software, model, and compositions in the series.

11:45

3aMU12. The acoustics of Fundação Iberê Camargo's parking lot. Peter Francis C. Gossweiler (Instituto de Artes, UFRGS, Trav. Nsa. Sra. de Lourdes, 230 apt. 503 D, Porto Alegre, Rio Grande do Sul 91920-040, Brazil, petergossweiler@gmail.com)

Through listening to the audio of a video recording of the acoustics of Fundação Iberê Camargo's parking lot, we propose the writing of a text. From this dependence on reliving past issues and reinterpreting them by a device such as this recording, we evoke the symptom of the void. To that end, we dialogue with Yves Klein's statement of le vide (1961), which proposes to feel and understand simultaneously the possibility of filling the space with our presence and absence, with the wooden doll's head Mechanischer Kopf (1920) by Raoul Hausmann, who, by dependence on multiple devices coupled to it, no longer exercises thinking and with Lygia Clark's Organic Line (1954), which makes us think that art is in the real world, just as the real world is in art. These relationships lead us to an organization of space for something beyond emptiness and our presence.

WEDNESDAY MORNING, 6 DECEMBER 2017 BALCONY L, 8:30 A.M. TO 11:45 A.M.

Session 3aPA

Physical Acoustics and Biomedical Acoustics: Acoustofluidics

Kedar C. Chitale, Cochair
Flodesign Sonics, 380 Main Street, Wilbraham, MA 01095

Max Denis, Cochair
U.S. Army Research Lab., 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Papers

8:30

3aPA1. Macro-scale acousto-fluidics using bulk ultrasonic standing waves. Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu), Kedar C. Chitale, Benjamin P. Ross-Johnsrud, Walter Presz (FloDesign Sonics, Wilbraham, MA), Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX)

Macro-scale acousto-fluidics involves the interaction between acoustic radiation force exerted on a particle by bulk acoustic standing waves spanning many wavelengths, fluid drag force, and the gravitational force of the particle. Parameters are particle size, and ratio of particle to fluid density and particle to fluid compressibility. Different acousto-fluidics configurations can be used to manipulate particles in multiple ways. In cell clarification, the configuration is that of a depth flow filter with the added benefit of separating the cells out of the acoustic field, thereby eliminating any issues with filter clogging or fouling. In perfusion of stirred bioreactors, the configuration resembles that of a tangential flow filter. In a third configuration, the bulk acoustic standing wave is angled relative to the fluid velocity resulting in a label-free fractionation tool. Several underlying theoretical and numerical results of acoustic radiation force and particle trajectory calculations will be presented. A theoretical framework to calculate the acoustic radiation force on spherical, spheroidal, and cylindrical particles has been developed for any particle size relative to wavelength. An analytical solution for particle deflection angle in a planar angled standing wave and uniform flow has been developed. Experimental results will be shown to support the theory.

9:00

3aPA2. Acoustic Rayleigh streaming: Comprehensive analysis of source terms and their evolution with acoustic level. Virginie Daru (Arts et Metiers ParisTech, Paris, France), Diana Baltean-Carlès, Catherine Weisman (Sorbonne Universités, UPMC Univ. Paris 6 and LIMSI-CNRS, Sorbonne Universités, UPMC Univ Paris 06, UFR d'Ingénierie, 4 Pl. Jussieu, LIMSI, CNRS, Université Paris-Saclay, Bât. 508, Rue John Von Neumann, Campus Universitaire, F-91405 Orsay Cedex, France, Paris 75252, France, baltean@limsi.fr), Hélène Bailliet (Institut Pprime, Poitiers, France), and Ida Reyt (Arts et Metiers ParisTech, PARIS, France)

Rayleigh streaming is a second order mean flow generated by the interaction between a standing wave and a solid wall. At moderate acoustic levels, the streaming flow is slow, composed of two cells along a quarter wavelength: an inner cell close to the tube wall and an outer cell in the core. When increasing the acoustic level, the streaming flow inside the inner cells is marginally modified, while the outer cells are strongly distorted. The emergence of an extra cell was observed both in previous numerical simulations and experiments and it has been shown that inertia is not responsible for this behavior, which is rather due to nonlinear interactions between streaming and acoustics. In the present work these interactions are analyzed both numerically and theoretically. The averaged Navier-Stokes equations are numerically solved with acoustic correlation source terms obtained from previous full instantaneous simulations. The effect of each source term is highlighted and the source term responsible for the emergence of the extra cell is identified. The numerical results are successfully compared with the analytical solution of simplified streaming linear equations.

9:30

3aPA3. Standing surface acoustic wave enabled acoustofluidics for bioparticle manipulation. Xiaoyun Ding (Mech. Eng., Univ. of Colorado at Boulder, 1111 Eng. Dr., ECES 158, Boulder, CO 80309, Xiaoyun.Ding@Colorado.edu)

Techniques that can noninvasively and dexterously manipulate cells and other bioparticles in a compact system are invaluable for many applications in life sciences and medicine. The past two decades have witnessed the emerging of acoustofluidics: the fusion of acoustics and microfluidics, due to it numerous advantages such as biocompatibility, miniaturization with lost cost, compatibility with other microfluidic system, and many others. Here, we first demonstrate surface acoustic wave (SAW) enabled acoustic tweezers platform for on-chip particle manipulation. Surface acoustic wave is a kind of ultrasound wave that propagates on the surface of a substrate. By tuning the standing SAW field, we demonstrated the functions of: (1) manipulation of nanoparticles, cells, and C elegance; (2) label free cell separation; (3) multichannel cell/droplet sorting, and (4) tunable cell patterning. Cells viability and proliferation assays were also conducted to confirm the non-invasiveness of our technique. The simple structure/setup of these acoustic tweezers can be integrated with a small radio-frequency power supply and basic electronics to function as a fully integrated, portable, and inexpensive cell-manipulation system. We believe that these unique advantages and functions position our acoustic tweezers to be a useful tool in medical diagnostics and biological/chemical studies.

Contributed Papers

10:00

3aPA4. Bacterial suspensions under acoustic confinement and the impact on biofilm formation. Salomé Gutiérrez (Laboratoire de physique et mécanique des Milieux Hétérogènes, ESPCI- Paris, France, UPMC, Université Paris7., 10 Rue Vauquelin, Paris, Ille de France 75005, France, salome.gutierrez-ramos@espci.fr)

Active matter exhibit plenty intriguing non equilibrium properties while in confinement. Optical and magnetic confinement had helped in the evaluation of rich collective behaviour of active entities. Self-propelled particles in aqueous suspensions can be trapped with acoustic fields generated by acoustic resonators usually at frequencies in the megaHertz. In this work, we give a step forward in the field of confined active matter reporting for the first time experiments of *Escherichia coli* suspensions in acoustic levitation. In particular the implementation of an acoustic levitation technique to manipulate in a contact-less and controlled way the dynamics of the active suspension. The aggregation of living bacteria is monitored as a function of time, where different phases are clearly distinguished. Upon the removal of the acoustic signal, bacteria rapidly disaggregate induced by their own swimming. However, if the levitation time increases, a stable aggregate, that resemble a biofilm, remains even after the withdraw of the acoustic confinement.

10:15–10:30 Break

10:30

3aPA5. Acoustic radiation force moment on non-spherical objects in liquid. Bart Lipkens (FloDesign Sonics, 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu), Yurii A. Ilinskii, and Evgenia A. Zabolotskaya (Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX)

Previously, a study of the acoustic radiation force acting on a spheroidal object in liquid showed that the radiation force depends on the angle between the incident acoustic wave and the main axis of a spheroid. This investigation demonstrated that there is a radiation force moment which acts on the spheroidal object and depends on particle orientation. This contribution is a continuation of the previously reported work. Here, the acoustic radiation force moment on prolate objects in an acoustic field in liquid is investigated analytically, i.e., the equations to describe the moment are derived. The incident acoustic and scattered field are expanded with respect to spherical waves. Analytically, scattering amplitudes are calculated from boundary conditions for spheroidal functions that are solutions of a wave equation in spheroidal coordinates ζ, η, φ. The radiation force moment is analyzed numerically. Randomly oriented spheroidal particles distributed in liquid align in an acoustic field as scattering amplitudes of each particle and therefore the acoustic radiation force acting on them depend on object orientation. The radiation force moments on each particle also depend on their orientation, and the prolate spheroidal objects are turned in such a way to make the torque of them to be equal to zero.

10:45

3aPA6. Fluid dynamical and acoustical fields in the vicinity of small objects reacting to radiation forces and torques. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), Charles Thompson (Elec. and Comput. Eng., Univ. of Massachusetts at Lowell, Lowell, MA), and Max Denis (Physical and Life Sci. Solutions, Lowell, MA)

The venerable and extensive literature on the theory of acoustic radiation forces and torques on small embedded particles is briefly reviewed, assessed, and criticized. This paper adopts a approach that first concentrates on determining the near field of a non-stationary embedded particle for the case when $ka<<1$. The near field is the region where $r<1/k$. The philosophy of matched asymptotic expansions is adopted and the near field is assumed to consist of (1) an oscillating incident wave, (2) an oscillating reaction wave that goes to zero at large r/a, (3) an oscillating incompressible field that is associated with viscosity, also going to zero at large r/a, and (4) a

quasi-static field that is also associated with viscosity. The latter results in part from a perturbation expansion to take into account non-linear effects of the fluid dynamical equations. The far-field scattered wave results from asymptotic matching to the near-field. The near-field reaction field separates into monopole and dipole portions where the orientation of the dipole is related the detailed natures of incident wave and particle. Forces are obtained by integration of stresses over the surface of the body or over a surface slightly outside the particle. In appropriate limits, the predicted forces agree with results of King (1934) and Gorkov (1962).

11:00

3aPA7. Investigation on a novel photoacoustofluidic effect. Gabriel P. Dumy, Mauricio Hoyos, and Jean-Luc Aider (PMMH, ESPCI Paris, 10 rue Vauquelin, Paris 75005, France, gabriel.dumy@espci.fr)

Acoustic manipulation of micro-objects (particles, cells, and bacteria) can be achieved using ultrasonic standing waves in a fluidic or microfluidic resonator. By matching resonator dimensions and acoustic field frequency, it is possible to use acoustic radiation force (ARF) to gather the particles in the pressure nodal (or anti-nodal) plane, creating one or several aggregates. In standard operating conditions, they can be maintained as long as needed in acoustic levitation at this equilibrium position. In this study, we present a new unexpected phenomenon. After creating a large aggregate of light-absorbing particles, we show that it is possible to force the complete breakup of the aggregate when we enlighten it with an electromagnetic wave of adequate wavelength and intensity. If the particles remain in acoustic levitation, they are quickly rejected and propelled away from the aggregate leading to its fast destruction. We show that this phenomenon strongly depends on both the amplitude of the ultrasonic field and the intensity of the lighting. Various experiments with different types of particles and concentrations are used to discuss the possible explanations of the phenomenon. Moreover, investigations showed that this phenomenon applies to biological compounds such as red blood cells and stem cells, suggesting potential biomedical applications.

11:15

3aPA8. Acoustical deformability of giant unilamellar vesicles. Liangfei Tian (School of Chemistry, Univ. of Bristol, Bristol, Bristol, United Kingdom), Glauber T. Silva (Dept. Phys., Federal Univ. of Alagoas, Av. Lourival Melo Mota, S/N, Maceio, Alagoas 57072-900, Brazil, tomaz.glauber@gmail.com), and Bruce W. Drinkwater (Dept. Mech. Eng., Univ. of Bristol, Bristol, United Kingdom)

An acoustic standing wave is used to trap and deform giant unilamellar vesicles with a diameter ranging from 10 to 50 μm. The giant unilamellar vesicles are prepared in glucose solution with a bi-layer of DOPC membrane with approximately 10 nm-thickness. They are suspended in a 4 cm^2-chamber of an acoustofluidic device. The density of the vesicles is about 98% of the external solution density. The device operates with a single-frequency at 6 MHz producing a standing wave of 250 μm-wavelength, which is much larger than the vesicles' radii. To explain the observed deformability, we propose an acoustic deformation model as follows. The radiation stress, caused by the interaction of the standing wave and a vesicle, is obtained in the long-wavelength limit. Using the deformation theory of thin spherical shells, we show that the aspect ratio of a deformed vesicle is $1 + 2\delta$, where δ is inversely proportional to Young's modulus and directly proportional to the density contrast between the vesicle and the solution. Our preliminary observations and theoretical results give an aspect ratio of the same order of magnitude, $\delta<1/2$. Additionally, predictions of our model agree with the results for the deformation of an osmotically swollen red blood cell reported by Mishra *et al.* [Biomicrofluidics **8**, 034109 (2014)]. In this case, the relative error is smaller than 6%. [Work partially supported by Newton Advanced Fellowship (NA160200), The Royal Society, UK.]

11:30

3aPA9. Dynamic measurement of blood viscoelasticity by an oscillatory acoustic tweezing technique. Nithya Kasireddy, Erika M. Chelales (Biomedical Eng., TulaneUniv., 440 Lindy Boggs, New Orleans, LA 70118, nkasired@tulane.edu), Vahideh Ansari Hosseinzadeh (Mech. Eng., Boston Univ., Boston, MA), Daishen Luo (Biomedical Eng., TulaneUniv., New Orleans, LA), Ray Holt (Mech. Eng., Boston Univ., Boston, MA), and Damir Khismatullin (Biomedical Eng., TulaneUniv., New Orleans, LA)

Acoustic tweezing rheometry is an innovative technology for low-volume non-contact rheological analysis of complex fluids characterized by increased sensitivity and accuracy as compared to traditional contact techniques. In this method, a small drop of a fluid sample is levitated in air by acoustic radiation forces and its viscoelasticity at different time instants is measured from drop shape changes. The acoustic tweezing rheometer operates in two different modes: quasi-static and oscillatory. This presentation focuses on the oscillatory technique in which the sample drop is forced into freely decaying shape oscillation by transient modulation of the standing acoustic field. Images of oscillating drops acquired by a high-speed camera are analyzed by a custom MATLAB code to obtain the shape amplitude vs. time curves and then the decay factor and resonance frequency of drop shape oscillation. Dynamic viscoelasticity of the sample (viscosity, relaxation time, and elastic modulus) was measured by applying the experimental data to the analytical formulae, derived from normal mode analysis of drop oscillation for viscoelastic fluid (Maxwell model) and viscoelastic solid (Kelvin-Voigt) materials. Using the oscillatory technique, we measured changes in blood viscoelasticity during coagulation and showed that the Kelvin-Voigt model leads to physically consistent results on rheology of coagulating blood.

Session 3aPPa

Psychological and Physiological Acoustics: Exploring the Perception of Sound (Poster Session)

Eric Hoover, Chair
University of South Florida, 16458 Northdale Oaks Dr., Tampa, FL 33624

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

3aPPa1. Effects of nonlinear frequency compression on Mandarin speech recognition and sound quality perception in hearing-aid users. Xueqing Chen, Yanyan You (Beijing Tongren Hospital, Capital Medical Univ.; Beijing Inst. of Otolaryngol., 17 HouGouHuTong, DongCheng District, Beijing 100005, China, xueqingchen2006@aliyun.com), Jing Yang (Speech and Hearing Sci., Univ. of Central Arkansas, Conway, AR), and Li Xu (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

The aim of the present study is to evaluate the effects of nonlinear frequency compression (NLFC) on Mandarin speech recognition and sound quality perception. Thirty Chinese-speaking, hearing-impaired adults without hearing-aid experiences participated in the study. They were fitted bilaterally with the Phonak behind-the-ear hearing aids. Each participant was evaluated with the Phoneme Perception Test, Mandarin speech recognition, and sound quality perception at post-fitting 0, 4, and 12 weeks with NLFC-on vs. NLFC-off. The NLFC were active at home. Results show significant differences in detection of phonemes of high-frequency components (such as /sa/) between NLFC on and off. The difference between NLFC on and off was significant for consonant and sentence recognition. Duration of NLFC use had significant effects on the recognition of consonants, vowels, and sentences. Regarding the sound quality perception test, there were significant effects of NLFC conditions (on vs. off), duration of NLFC use, and sound types on the four categories of percepts (i.e., loudness, clarity, naturalness, and overall preference). Therefore, NLFC improves the audibility of high-frequency speech signals and provides better recognition of consonants and sentences and perception of sound quality. The benefit of NLFC on speech recognition and sound quality perception requires certain amount of acclimatization.

3aPPa2. Sensitivity of pure tone versus speech-in-noise hearing screening. Lawrence L. Feth, Evelyn M. Hoglund, Christina M. Roup, and Kaitlin Campbell (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, feth.1@osu.edu)

Adults classified as having normal hearing using conventional pure tone hearing screening often report hearing difficulties especially for situations requiring listening for speech or music in a background of noise. The purpose of this project is to compare the sensitivity of a pure tone hearing screener with a Spoken Digits in Noise test. In the first comparison, 20 young adults with normal hearing thresholds documented by a full audiometric test were asked to simulate a conductive hearing loss using an ear plug inserted into one ear. Both screeners were used to detect the simulated conductive hearing loss. In the second comparison, participants with documented mild to moderate sensorineural hearing loss were tested with both screeners. Order of testing with the screeners was counter-balanced, and the degree of hearing loss was categorized into 15 dB intervals based on their three-frequency (500, 1000, and 2000 Hz) averages and labeled Normal, Slight, Mild, or Moderate. A decision theory analysis was used to indicate the sensitivity of the two screening procedures. [Work supported by a grant from NIDCD.]

3aPPa3. Does the right ear advantage persist in mature auditory systems when cognitive demand for processing increases? Danielle M. Sacchinelli (Commun. Disord., Auburn Univ., 67 Waverly Ave., Eastchester, NY 10709, dms0043@auburn.edu), Aurora J. Weaver, and Martha Wilson (Commun. Disord., Auburn Univ., Auburn, AL)

Based on Kimura's (1967) anatomical model of dichotic listening, the right ear has a slight advantage (REA) or performance asymmetry, compared with the left ear. This is due to left hemisphere dominance for language, which receives direct input from the right ear. Accurate performance on dichotic tests relies on sensory organization and memory; however, there is little evidence regarding the impact of increasing cognitive demands (i.e., number of items for recall) on auditory performance asymmetries in mature auditory systems. This study investigated 42 participants' auditory perceptual and working memory abilities (e.g., forward and reverse digit spans, dichotic digits test, and directed ear dichotic digits test) to explore the relationships among cognitive demanded and performance asymmetries. Repeated measures ANOVA showed a significant effect for directed ear and digit list length. In addition, a priori comparison indicated significant performance asymmetry, with persistent REA when listening demands exceeded an individual's auditory memory capacity. No significant performance differences were identified for digit list lengths relative to, or below an individual's simple memory capacity. Overall, the study found the right ear tends to show better performance on dichotic listening tasks, even in adults, when the number of digits exceeded the participants' digit span capacity.

3aPPa4. Development and validation of a portable platform for auditory testing. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Aaron Seitz, Trevor Stavropoulos (Univ. of California, Riverside, Riverside, CA), David A. Eddins, Eric Hoover (Univ. of South Florida, Tampa, FL), Samuel Gordon, Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), Kasey Jakien, and Anna Diedesch (Oregon Health & Sci. Univ., Portland, OR)

This presentation will describe a portable platform for psychoacoustical testing using relatively inexpensive components that can be used outside of the laboratory, making it available for a broad range of research, clinical, and academic uses. This project is based on the concept of leveraging the fact that, due to the support of the tools and community of the $100 billion gaming industry, high-performance interactive audio-visual systems are now easily available in the form of consumer devices. The first task was to develop mobile applications capable of administering tests of multiple psychoacoustics tests such as (1) binaural processing, (2) spectrotemporal sensitivity, and (3) speech intelligibility in competition with and without spatial cues. The second task was to verify that the system performed within design specifications, by acoustic analysis of the sound output and comparison of behavioral thresholds to those obtained using traditional methods. The final task was the creation of experimenter interfaces that facilitate the use of the system by researchers and clinicians without requiring expertise in the

underlying technologies. Data and examples of our success with all three tasks will be presented, implemented with a system based on only two relatively inexpensive devices: an Apple iPad Pro with audio output connected directly to Sennheiser HD 280 Pro headphones. [Funding provided by NIH R01 DC 015051 and VARR&D NCRAR.]

3aPPa5. Effects of residual masker, spectral resolution, and low frequency cues on the perception of ideal binary-masked speech. Vahid Montazeri and Peter F. Assmann (GR 41, School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX 75080, vahid.montazeri@utdallas.edu)

This study investigated the intelligibility of ideal binary-masked (IdBM) stimuli under conditions with limited spectral resolution. In experiment 1, we used different IdBM local thresholds, which retain different amounts of residual masker and target information in the IdBM-stimuli. The stimuli were presented to 30 normal hearing listeners. The results indicated that, with a 6- or 12-channel tone-vocoder, the presence of residual masker in the IdBM-stimuli limits the intelligibility scores, thus preventing IdBM processing from achieving an ideal segregation of target from masker. In experiment 2, we investigated whether introduction of low frequency target information to the tone-vocoded IdBM-stimuli improves the intelligibility scores. Twenty normal hearing listeners participated in this experiment. The results indicated that in contrast to target F0 cues, inclusion of low-pass filtered target information (cutoff = 300 Hz) helped listeners segregate the target from the masker, thus improving the intelligibility of the IdBM-stimuli. These results argue against F0 as a segregation cue in electroacoustic conditions and suggest that target F0 cues do not provide benefits beyond those provided by IdBM.

3aPPa6. Spectrum contrast sharpening by lateral suppression: Comb-filtered noise test. Alexander Supin, Dmitry Nechaev (Institute of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru), Vladimir Popov, and Evgeniya Sysueva (Institute of Ecology and Evolution, Moscow, Russian Federation)

Spectrum contrast thresholds were measured in normal listeners using comb-filtered signals. The signal spectrum contained 0.075-oct wide ripples. Spectrum contrast varied from 0 (no ripple on the pedestal) to 1 (zero spectrum level between the ripples). Spectrum contrast thresholds were measured at ripple densities from 1 to 10 oct^{-1} using a two-alternative forced-choice adaptive procedure. The lowest spectrum contrast thresholds of less than 0.1 appeared at a ripple density of 4 oct^{-1}. At lower ripple density down to 1 ocn^{-1}, thresholds increase up to 0.2. At ripple densities above 4 oct^{-1}, thresholds steeply increased up to 1.0 at densities of 8 to 10 oct^{-1}. In agreement with the excitation-pattern model, threshold increase at high ripple densities may be explained by across-frequency integration in critical bands. However, threshold increase at low ripple densities cannot be explained by a simple critical-band model. The lowest spectrum contrast thresholds at a certain inter-ripple interval indicate the presence of lateral suppression zones in equivalent filter forms. According to this interpretation, the results demonstrate sharpening of spectral contrast due to lateral suppression. The lateral suppression zones may reflect either cochlear lateral suppression, or lateral inhibition in neuronal centers, or both. [Work supported by Russian Science Foundation.]

3aPPa7. Predictive denoising of speech in noise using deep neural networks. Manuel Pariente and Daniel Pressnitzer (Laboratoire des Systèmes Perceptifs, CNRS, ENS, PSL, 29 Rue d'Ulm, Paris 75005, France, pariente.mnl@gmail.com)

Deep neural networks have recently been used in several studies to improve speech intelligibility in noise. Here, we tested whether such networks could denoise speech in a predictive manner, which would be highly desirable for potential real-time applications. Training targets for the networks consisted in a mask (ideal binary mask or ideal ratio mask) for the last observed frame (non-predictive) and for one frame ahead in the future (predictive). Frame length was fixed at 48 ms. Training was performed on a target speaker with added speech-shaped noise, using about 25 min of training speech, at different signal to noise ratios ranging from −12 to 3 dB. A behavioral experiment was run to measure intelligibility of semantically unpredictable sentences in speech-shaped noise. For the behavioral experiment, target sentences were different from the learning sentences, and novel exemplars of speech-shaped noise were drawn on each trial. We observed intelligibility gains for both network architectures over a broad range of signal to noise ratios, with a maximum of 13.6 percentage points for the non-predictive network compared to 9.4 percentage points for the predictive network. This shows that a network may successfully be trained to denoise a specific speaker in a predictive manner.

3aPPa8. Pure-tone lateralization revisited. Florian Völk (WindAcoust., Muehlbachstrasse 1, Windach 86949, Germany, voelk@windacoustics.com), Jörg Encke, Jasmin Kreh, and Werner Hemmert (Bio-Inspired Information Processing, Tech. Univ. of Munich, Garching, Germany)

In certain conditions, especially with diotic headphone presentation, hearing sensations are located inside the head. Dichotic headphone presentation can result in lateralization: delaying one of the headphone input signals or reducing its amplitude typically pushes the hearing sensation inside the head toward the contralateral ear. Systematic connections between physical stimulus parameters and hearing-sensation positions provide insight into auditory-localization mechanisms and are helpful in designing and evaluating models thereof. However, typical paradigms of addressing lateral displacement, magnitude estimation or pointing, may suffer from response biases. This study aims at evaluating the suitability of a two-alternative forced choice paradigm: 15 normal-hearing subjects were asked to indicate, by pushing one of two buttons, whether the hearing sensation associated with a pure interaural phase or amplitude difference occurred "left or right," without any reference or further instructions. Posing a notably simple task without internal mapping, this procedure appears advantageous regarding response biases. The results indicate a high intra-individual reproducibility and plausible inter-individual agreement. None of the participants encountered or reported difficulties using the inherently assumed "internal center" in their decision process. The data suggest the existence of an inter-individually similar decision criterion. Descriptively speaking, the results support the existence of a similar lateralization midpoint.

Session 3aPPb

Psychological and Physiological Acoustics: Perception and Physiology: Musicians, Musical Instruments, and the Body (Poster Session)

Eric Hoover, Chair
University of South Florida, 16458 Northdale Oaks Dr., Tampa, FL 33624

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

3aPPb1. A study on auditory sensitivity and behavioral cognition changes before and after drinking in psychological acoustics. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Korea, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and Myungjin Bae (Information and Commun., Soongsil, Seoul, South Korea)

In general, an auditory sensitivity after drinking is becoming insensitive and a perception of behavior becomes very poor. This can be explained by the characteristics of body changes due to alcohol absorption, and this study analyzed and studied these characteristics. This study provides the necessary bases for psychoacoustics and brain cognitive science by obtaining a measured data through cognitive testing of the characteristics of body changes before and after drinking. We analyzed the post-drinking changes in brain waves and performed psychoacoustic analysis of these characteristics.

3aPPb2. Relationship between behavioral and electrophysiological hearing thresholds. Edward L. Goshorn, Charles G. Marx, and Kimberly Ward (Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, PsychoAcoust. Res. Lab., Hattiesburg, MS 39401, edward.goshorn@usm.edu)

Clinical applications for hearing thresholds obtained with electrophysiological measures are well established. Bush, Jones, and Shinn (2008) suggested that a diagnostic relationship exists between behavioral and auditory brainstem response (ABR) thresholds. They reported that subjects with MRI-confirmed vestibular schwannomas had differences greater than 30 dB when behavioral thresholds were compared to ABR using a 100 microsecond square wave (click) stimulus. However, there are little data available showing threshold relationships between behavioral and ABR thresholds for frequency specific stimuli. It is conceivable that a frequency specific stimulus may be more sensitive to schwannoma effects. Therefore, this project examined differences in hearing thresholds for normal hearing participants for two threshold methods (behavioral and ABR) and four stimuli: 250, 1000, and 4000 Hz tone bursts, and a 100 microsecond click. The same ABR instrument and stimuli were used for each method to obtain thresholds in twenty eight normal hearing adults in a repeated measures design. Results showed ABR thresholds to be significantly higher than behavioral. Thresholds varied significantly across stimuli for the ABR method but not the behavioral. Differences between methods were less than 30 dB for 93% of participants. These findings are consistent with Bush's findings for a normal hearing control group.

3aPPb3. Electrophysiological determination of backward masking function. Silas Smith and Al Yonovitz (Dept. of Communicative Sci. & Disord., Univ. of Montana, Missoula, MT 59812, silas.smith@umontana.edu)

Backward Masking (BM) functions have been shown to relate to age, lead toxicity, and in children with language disorders. These functions may be indicative of auditory processing deficits. This study investigated if Evoked Potentials (EP) could be utilized to obtain BM functions. The design of this study allowed observation of the early, middle, and late auditory evoked potentials. This process allowed us to observe the differential electrophysiological responses of evoked potentials during the BM effect. This study was randomized using four different stimulus conditions. With a long inter-stimulus interval and high sample rate simultaneous early, middle, and late potentials were obtained. The stimuli were a pure-tone alone, (1) 1000 Hz, (2) a masking noise alone, (3) the pure-tone followed by the masking noise, and (4) a control condition with no auditory stimulus. This approach, using a randomization allowed any adaptation or habituation to the stimuli to be equally distributed within each condition. Using arithmetic operations on the derived evoked potentials allows for localization of brain structures utilized in BM processing. Results indicated the determination of BM functions can be obtained objectively.

3aPPb4. Crowdsourcing the creation of an audio dataset for human and machine medical diagnosis training. Scott H. Hawley (Chemistry & Physics, Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu), Tamara Baird (School of Nursing, Lipscomb Univ., Nashville, TN), and Frank Baird (Recording and Entertainment, Middle Tennessee State Univ., Murfreesboro, TN)

We describe a method for creating and maintaining an open, annotated, community-moderated dataset of audio recordings of heart and lung sounds, with which to train machine listening systems to perform medical diagnosis. This is achieved by partnering with education programs for nursing and medical professionals who will receive training in diagnosis using digital stethoscopes. We developed a low-cost digital stethoscope using a peer-reviewed, open-source, 3-D printed design. With sufficiently numerous examples supplied and tagged by nursing and medical students, it is possible to employ machine learning classifiers such as our convolutional neural network code—originally developed for music information retrieval—to identify diagnosis classes. While primary intent of this dataset-creation and moderation system is for medical audio, the underlying functionality of community moderation could be applied to other waveform content, including a variety of musical datasets.

3aPPb5. Evaluation of reproduction fidelity of acoustic characteristics of auscultation sounds by recordings included in various publicly available databases. Karolina M. Nowak (Dept. of Endocrinology, Ctr. of Postgraduate Medical Education, Ceglowska 80 St., Warsaw 01-809, Poland, karolina.brodowska@gmail.com) and Lukasz Nowak (Dept. of Intelligent Technologies, Inst. of Fundamental Technol. Res. Polish Acad. of Sci., Warszawa, Poland)

Various databases of auscultation sounds recordings are available for teaching purposes for physicians. The sounds included in these databases were recorded using different equipment, and—in some cases—generated artificially. Thus, an important question arises regarding the differences in acoustic parameters between the recordings of bioacoustic signals obtained using different techniques and hardware, and the sounds heard by a physician during an actual patient auscultation through a typical acoustic stethoscope. The acoustic stethoscopes are the most widespread diagnostic devices and are far more popular in clinical practice than the electronic ones. The present study introduces the results of an analysis of acoustic characteristics of sounds included in various databases, compared to the parameters of sounds recorded with a microphone placed in an earpiece of an acoustic stethoscope. It is shown that the differences in time- and frequency characteristics are large enough to question the legitimacy of using various databases for teaching purposes. An alternative method and setup for recording bioacousic signals that could be used for developing auscultation skills is introduced.

3aPPb6. Assessment of acoustic characteristics of the nasal cavity by measuring sound transmission from one nostril to the other by a small speaker and a microphone. Amitava Biswas (Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Sometimes clinicians need to assess aerodynamic and acoustic characteristics of the nasal cavity. Some patients with cranio-facial anomalies such as cleft palates may need such assessments. Certain disease specific pathologies may involve temporary blockage in the nasal cavity. This study will present a simple procedure of computing the transfer function of acoustic energy from one nostril to the other.

3aPPb7. Relationships among musical training and pitch matching in children and adults. Aurora J. Weaver (Commun. Disord., Auburn Univ., 1139-C Haley Ctr., Auburn, AL 36849, ajw0055@auburn.edu), Jeffrey J. DiGiovanni (Commun. Sci. and Disord., Ohio Univ., Athens, OH), and Dennis Ries (Commun. Sci. and Disord., Fort Hays State Univ., Athens, OH)

This experiment attempted to determine if individuals with extensive musical training's pitch perception and memory were more resistant to degradation (e.g., time and interference) than that of individuals with limited musical training. It is known that musical training influences cortical sound processing through learning-based processes, but also at the preattentive level within the brainstem. Pitch memory abilities were investigated in 66 participants with no known hearing, attention, or cognitive impairment. Participants were placed into subgroups based on age (young children, older children, and adults) and their self-reported musical training experience. Two experiments measuring auditory perception and memory skills for pitch were collected, the pitch pattern span (PPS; Weaver, DiGiovanni, & Ries, 2015) and a pitch matching retention task based on Ross, Olsen, and Gore's procedure (2003). We found that individuals with greater musical training exhibited enhanced pitch perception and memory processes and smaller *standard deviation* across pitch matches. Unexpectedly, based on the paradigm, the young children demonstrated significantly sharper (higher) *constant error* across pitch matches than the older participants. These and additional findings will be discussed with reference to task parameters which attempted to remove the typically required knowledge of musical nomenclature in pitch matching tasks

3aPPb8. Patterns procedural learning for pitch matching in adult musicians and non-musicians. Megan M. Barnett, Aurora J. Weaver, and Anne Rankin Cannon (Commun. Disord., Auburn Univ., 1199 Haley Ctr., Auburn, AL 36849, mmb0022@auburn.edu)

This study evaluated the role of active musical training on procedural learning during a pitch perception, memory, and matching task. Twenty-one adults with hearing within normal limits were split into music subgroups (musician and non-musicians) based on active participation in musical training. Ninety pitch-matching trials were completed using a paradigm adapted from Ross, Olsen, and Gore (2003) which assesses pitch-matching precision without requiring knowledge of musical nomenclature. Three blocks of 30 pitch-match trials were collected to evaluate the learning effect on each music subgroup. The semitone half step (HS) distance between the target pitch and the comparison pitch match was calculated for each trial. Constant error (*CE*) and standard deviation (*SD*) were used to quantify performance across blocks. Overall, significant reduction of pitch match *SD* was found for both music subgroups across blocks. Musicians demonstrated more precise pitch matches (smaller *SD*) and demonstrated no significant changes in pitch matching *CE* across blocks, whereas non-musicians *CE* was reduced across blocks. The pattern of results indicates learning effects, which we attribute to procedural memory for both groups, based on task parameters, while the music group's performance across blocks may reflect enhanced selective attention and pitch perception.

3aPPb9. Musical instrument dexterity requirements and its effects on adolescent diotic and dichotic listening performance. Aurora J. Weaver (Commun. Disord., Auburn Univ., 1139-C Haley Ctr., Auburn, AL 36849, ajw0055@auburn.edu), Naveen K. Nagaraj (Audiol. and Speech Pathol., Univ. of Arkansas for Medical Sci., Little Rock, AR), and Abby N. Turnbough (Commun. Disord., Auburn Univ., Auburn, AL)

Research examining the human asymmetry of handedness has included exploring the relationship to the auditory modality (e.g., digit span memory; Bannatyne & Wichiarajote, 1969). Musicians whose instruments require dual dexterity have more symmetric neural processing as a result of the sensory-motor experience with their instrument (Gaser & Schlaug, 2003). Little research translates these neurological differences based on dexterity to determine if more neurologic symmetry manifests into advantages for auditory processing. This study aimed to identify perceptual advantages of instrumental training based on dexterity, during stages when the central auditory nervous system is still developing. Monaural and binaural listening tasks, as well as working memory tasks, were collected on 33 adolescent (14–18 years old) musicians split into subgroups based on their instrument dexterity requirement (mono vs. dual). The outcomes indicate that instrumental choice did not significantly impact diotic digit span performance; however, a performance asymmetry (e.g., right ear advantage) was identified for the mono dexterity group for dichotic listening tasks performance at capacity limits. Overall, the results indicate that individuals choosing musical instruments that require dual dexterity had more symmetric auditory processing which corresponds to previous evidence of more neural symmetry based on instrumental dexterity.

3aPPb10. Analysis of singing bowl's sound. Ik-Soo Ann (Cultural Contents, Soongsil Univ., 369 sangdo-ro, Dongjak-gu, Seoul, Seoul 156-743, South Korea, aisbestman@naver.com) and Myungjin Bae (Information and TeleCommun., Soongsil Univ., Seoul, South Korea)

Studies about the effects of sound on the human body have been proven by various methods. Among them, we researched the sound healing tool known as singing bowl. The singing bowl is a bowl made of bronze and tin. It varies in size and thickness. One or several different singing bowls are placed either on or around the human body. The bowls are then rubbed or knocked using a small rod to produce sound. A singing bowl generates different frequencies depending on its size and thickness. The frequency vibrates the flesh and bone as well as the brain and organs of the human body. By doing so, it recovers the natural frequency of each part of the human body. Finally, we researched how the frequency generates an average frequency of some form as well as how it reacts to the human body through a brain waves analysis and a frequency analysis of the singing bowl sound.

3aPPb11. Sound art with Hertz. Peter Francis C. Gossweiler (Instituto de Artes, UFRGS, Trav. Nsa. Sra. de Lourdes, 230 apt. 503 D, Porto Alegre, Rio Grande do Sul 91920-040, Brazil, petergossweiler@gmail.com)

This article seeks to reflect on personal artistic processes in sound art and acoustic art pieces. It reports experiences, work pieces, and exhibitions where I applied a conventional language of sound measurement (Hertz) to minimize unexpected narratives and interpretations. Finally, we present the exhibition "HERTZ of the place where we are in" (2016), which has been audiodescripted for blind people, and thus we understand a little more of how we can fill these distances between language and its referent with alternative approaches and how we think about what we hear in acoustic environments.

WEDNESDAY MORNING, 6 DECEMBER 2017 BALCONY N, 8:30 A.M. TO 11:05 A.M.

Session 3aSA

Structural Acoustics and Vibration: Applications of Finite Element Analysis, Boundary Element Analysis, and Statistical Element Analysis Computational Methods

Elizabeth A. Magliula, Cochair
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James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Chair's Introduction—8:30

Invited Papers

8:35

3aSA1. Simplified shell modeling for submerged structures. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

Faithful numerical models of the acoustical response of submerged structures are complicated by the necessity to account for the interaction of three fundamentally different systems: the outer pressure hull, the internal substructure, and the surrounding fluid. Standard finite element techniques are the only recourse for a complex substructure, and boundary element or finite element techniques usually are necessary for the fluid. This paper proposes an alternative approach for the pressure hull based on an observation regarding the relative importance of extensional and flexural effects in a curved shell. The formulation entails applying an energy-based correction to membrane theory in order to account for flexural deformation. In some cases doing so enables an analytical representation of the shell's behavior, and it also simplifies a finite element representation. Whereas the interaction laws between the shell and the fluid are enforced conventionally, coupling of the shell and the substructures may be implemented a procedure rooted in analytical dynamics. It imposes constraint equations on the displacement variables, which introduces Lagrange multipliers to the equations of motion. In this manner, explicit consideration of the forces at attachments is avoided.

8:55

3aSA2. Investigation of variance response in energy finite element analysis. Kuangcheng Wu (Naval Surface Warfare Ctr. - Carderock, 9500 MacArthur Blvd., West Bethesda, MD 20817, kuangcheng.wu@navy.mil) and Nickolas Vlahopoulos (Univ. of Michigan, Ann Arbor, MI)

Noise and vibration control are important design features in various industries. Energy Finite Element Analysis (EFEA) and Statistical Energy Analysis (SEA) have been widely used in analyzing realistic structural and acoustic system for mid-to-high frequency regime. The EFEA is derived from the wave equation and uses energy density as its primary variable in its governing equation which is numerically solved in finite element approach. Generally, the EFEA calculates the mean values of given vibro-acoustic systems. This research introduces variance bounds into the EFEA analysis. Specifically, the variance originated from input power and joint matrix is incorporated to the EFEA. The formulation for calculation of the variance in the EFEA will be discussed and followed by several benchmark examples.

9:15

3aSA3. Adaptive manifold interpolation techniques for acoustic metamaterial characterization in HPC environments. Corbin Robeck (Weidlinger Appl. Sci., Thorton Thomasetti, 2000 L St. NW Ste. 600, Washington, DC 20036, CRobeck@ThorntonTomasetti.com), Heather Reed, Alex Kelly, Andrew Shakalis (Weidlinger Appl. Sci., Thorton Thomasetti, New York, NY), Reza Salari (Weidlinger Appl. Sci., Thorton Thomasetti, Cupertino, CA), and Jeffrey Cipolla (Weidlinger Appl. Sci., Thorton Thomasetti, Washington, DC)

Functionally graded acoustic metamaterials (FGAMs) can be designed to have specific waveguide properties dictated by a theory relevant to the application. Frequently, these material properties do not exist naturally, and must be fabricated by gradually layering manufactured unit cell microstructures, resulting in a (usually smooth) variation of properties. Tailoring these microstructures to the demands of the relevant theory requires assembling microstructures with very specific material properties—a process that often requires haphazardly searching a large domain space of unit cells for desirable parameter combinations. Moreover, the process of determining the material tensor of interest for a specific set of metamaterial cell design parameters typically involves solving a costly high dimensional finite element problem for each new microstructure cell of interest. This work presents a gradient-based, manifold interpolation technique to characterize acoustic metamaterial homogenized elastic tensors. An adaptive refinement method is used to seed the interpolation data allowing the method to reliably recover unit cell stiffness tensors within predefined error bounds—allowing for interpolation of a cell with desired properties within specified error tolerances while minimizing the number of unit cells that require high-fidelity, computationally-expensive simulation. Finally, an implementation of the process is presented in a massively parallel computing environment.

9:35

3aSA4. The effect of inertial distribution on dispersive modes in pentamode metamaterials for use in acoustic cloaking. Alex Kelly, Andrew Shakalis (Weidlinger Appl. Sci., Thorton Thomasetti, New York, NY), Reza Salari (Weidlinger Appl. Sci., Thorton Thomasetti, Cupertino, CA), Corbin Robeck (Weidlinger Appl. Sci., Thorton Thomasetti, Washington, DC), Heather Reed (Weidlinger Appl. Sci., Thorton Thomasetti, New York, NY), and Jeffrey Cipolla (Weidlinger Appl. Sci., Thorton Thomasetti, 2000 L St. NW Ste. 600, Washington, DC 20036, JCipolla@ThorntonTomasetti.com)

Transformation acoustics uses invertible maps to transform a cloaked region to a region with a smaller scatterer. The "elastic" or Norris class of acoustic cloaking theory is inherently broadband, in that it does not rely on the presence of resonant effects. To achieve the properties required by the transformation theory, we utilize pentamode materials, which admit the use of finite mass but require anisotropic stiffness throughout the material. These types of materials do not exist naturally and must be fabricated through the use of metamaterials. One of the challenges associated with metamaterials for use in dynamic applications is the distribution of mass in a unit cell and its consequences for frequency-dependent behavior in the metamaterial. Various homogenization techniques for recovering wave speed properties of the metamaterial unit cells may predict contradictory wave speed values for pentamodal structures. These disagreements can be explained through the distribution of inertia in the material and the resulting proliferation of dispersive modes present in the Bloch-Floquet diagrams. This effect will be described within the context of static homogenization, Bloch-Floquet theory, and numerical experiments effected through finite element software PZFlex. An approach for optimization of these metamaterials with respect to the dispersive modes will be discussed.

9:55–10:15 Break

10:15

3aSA5. Microstructure-in-the-loop optimization of acoustic metamaterial structures. Reza Salari (Weidlinger Appl. Sci., Thorton Thomasetti, 19200 Stevens Creek Blvd., Ste. 100, Cupertino, CA, MSalari@ThorntonTomasetti.com), Corbin Robeck (Weidlinger Appl. Sci., Thorton Thomasetti, Washington, DC), Alex Kelly, Heather Reed, Andrew Shakalis (Weidlinger Appl. Sci., Thorton Thomasetti, New York, NY), and Jeffrey Cipolla (Weidlinger Appl. Sci., Thorton Thomasetti, Washington, DC)

The field of acoustic metamaterial research has been driven, first, by transformation acoustics cloaking theory. This and other theoretical approaches optimize an acoustic effect through definition of artificial material properties which at present are only achievable using metamaterial technology. Metamaterials, however, present separate challenges in optimization for any desired acoustic effect: in particular, their dynamic behavior depends on parameters unrelated to acoustics, and exhibits wave propagation behavior more complex than elastic or acoustic continua. Homogenization methods which assume Cartesian symmetry are a staple in metamaterial design, but these only approximate a feasible optimal design. Moreover, total reliance on homogenized continuum models provides no information about the actual microstructure performance, and presents problems in functionally graded applications. To overcome this, we augment our Cartesian homogenization processes with high-resolution finite element models to optimize the design. Comparatively computationally expensive implicit FEM is avoided; specifically tailored time-domain wave propagation codes make the analyses feasible. Examples of the process, combining Cartesian homogenization estimates with high-resolution microstructural wave propagation solutions, will be shown.

Contributed Papers

10:35

3aSA6. Characterizing hysteretic materials in complex systems from vibration measurements. Alyssa T. Liem and James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, atliem@bu.edu)

A new method is proposed for estimating the material properties of a component in a complex system, given vibration measurements taken at points on the system. The method begins by identifying a set of unknown material parameters for each material. A finite element model is constructed using initial estimates of these parameters. A set of error metrics is defined and each metric is assumed to be zero when the correct parameters are used in the model. These error metrics may, for example, include averages of vibrational responses or modal properties. By evaluating the finite element model as the material parameters are varied, relationships between the material parameters and the error metrics are established. The best estimates of the material parameters are found by requiring that all error metrics be zero. The method is particularly valuable when applied to the *in situ* determination of hysteretic material properties, in which the frequency-dependent

constitutive law might contain several unknown parameters. Examples will be presented that illustrate the accuracy and robustness of the method.

10:50

3aSA7. Efficient prediction of the transmission loss of curved systems with attached noise control treatment. Kamal Kesour and Noureddine Atalla (GAUS Mech. Eng., Univ. of Sherbrooke, UdeS, 2500, boulevard de l'Université, Sherbrooke, QC J1K2R1, Canada, kamal.kesour@usherbrooke.ca)

This paper discusses the modeling of the transmission loss of curved panels with attached sound absorbing materials (foam or fiber). Two approaches are compared. The first is a classical coupled FEM/BEM (Finite element/Boundary element) approach wherein the panel and its cavity are modeled using the FEM and the blocked pressure on the excitation side and radiation from the transmission hole are modeled using the BEM. The second is an approximate approach using a Patch Transfer Function (PTF) method to describe the coupling between the panel, the sound package, the cavity and the transmission hole. To compute the PTF relations, the panel and the cavity are described by FEM while the effect of the sound package is described by a Green's function methodology using the Transfer Matrix Method (TMM) to speed up the process. It is shown that this latter approach is quick and accurate enough to estimate the TL of the system. Moreover, it allows for a quick comparison of the performance of various sound packages for a given curved panel. Several examples are presented to demonstrate the validity of this approximation and demonstrate its range of applicability and usefulness.

WEDNESDAY MORNING, 6 DECEMBER 2017 — ACADIA, 8:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication and Education in Acoustics: Teaching Phonetics and Speech Science in the New Millennium: Challenges and Opportunities

Catherine L. Rogers, Cochair
Dept. of Communication Sciences and Disorders, University of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Stories of speech science. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

A fundamental aspect of teaching, on any topic, is the continual pursuit of telling a story. Although technology and advances in teaching methods may facilitate new and exciting forms of presenting course materials, they do not, by themselves, build the context for the content of a course. Every lecture, activity, homework assignment, project, quiz, and examination can be regarded as chapters that build, over the duration of a course, a compelling and engaging story in which students take part. The aim of this talk is to encourage development of speech science courses that weave together history, theory, technology, visual and auditory experience, assessment, and, importantly, the instructor's own research to spin a good tale. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

8:25

3aSC2. Teaching phonetics and speech science—What's changed and what hasn't. Fredericka Bell-Berti (Commun. Sci. and Disord., St. John's Univ., 8000 Utopia Parkway, Queens, NY 11439, Fbellberti@gmail.com)

Many topics in articulatory phonetics that have been taught with traditional pedagogical techniques may be enhanced by the addition of acoustic information. However, it can be challenging to provide that information in ways that are accessible to students of Speech and Hearing, who often lack a strong background in physics. The ASA Committee on Education in Acoustics has offered sessions on acoustics demonstrations for ASA members. In meeting cities, the committee has hosted sessions for high school students and has co-sponsored sessions for Girl Scouts with Women in Acoustics. These sessions have resulted in a number of straightforward, accessible demonstrations of acoustics that are appropriate for use in undergraduate phonetics and speech science courses. This presentation will describe several traditional and computer-based demonstrations that may be used to teach speech acoustics to undergraduates in Speech and Hearing.

8:45

3aSC3. Some thoughts on teaching and learning phonetic transcription. James Hillenbrand (Western Michigan Univ., 1903 W Michigan Ave., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

Phonetics instructors often observe considerable variability in the ease with which students learn phonetic transcription. As an initial step toward understanding this variability, 50 students were tested on 12 phonological awareness (PA) tasks (e.g., finding the odd vowel, odd consonant, or odd stress pattern in a set of words, determining the word that would result from speech sound substitution, deletion, or reversal, …). Students were tested at the beginning of an introductory phonetics course. PA performance was then compared with student performance on: (1) quizzes and exam items related to transcription, and (2) student performance on all aspects of the course other than transcription. A strong relationship ($\rho = 0.84$) was found between average PA and transcription performance. However, a very strong relationship ($\rho = 0.88$) was also found between average PA and student performance in all areas of the course other than transcription. This finding suggests the possibility of underlying causal relationships that are considerably more complicated—and more general—than a simple dependency on PA skills. In more practical matters, the talk will also describe a collection of computer exercises that were developed to provide students with drill in phonetic transcription.

9:05

3aSC4. Teaching an online phonetics course: One approach. Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Ohio State offers an on-line graduate-level phonetics course for a three-year on-line MA program in speech-language pathology. It is taught using a combination of available power-point presentations (with embedded lectures), high-quality sound files (for transcription exercises), and weekly video chats (using carmenconnect). The course syllabus is an on-line syllabus (in.html format) created using Adobe Dreamweaver. There are links in the syllabus to all readings and links to useful educational sites for description of speech anatomy and physiology, vocal cord physiology (and dynamics of voicing), x-ray cinematography, and acoustic analysis of speech production. In terms of phonetic transcription, students move from relatively easy consonant and vowel transcription (using recorded nonsense words) through narrow transcription using a series of graded downloadable transcription exercises. Students download free phonetic fonts from the internet (and they can use on-line browser-based programs like ipa.typeit). The final two-three weeks of the course involves spectrographic analysis (using programs that are free online including PRAAT, Speech Analyzer 3.1, and Wasp). One of the more challenging issues is to determine programs that will work on both Windows and/or Apple computers) and the first week is dedicated to addressing technology issues. Examples of all materials will be presented.

9:25

3aSC5. Phonetics online. How far is self-study possible? Patricia Ashby (English, Linguist and Cultural Studies, ELCS, Univ. of Westminster, 32-38 Wells St., London WC1T 3UW, United Kingdom, ashbyp@westminster.ac.uk)

Using the *Certificate* examination of the International Phonetic Association as a benchmark, this talk explores the viability of computer-assisted self-study in phonetics. Historically, phonetics and technology have always gone hand in hand. From the 18th century, invention and application of specialist instruments has expanded our knowledge of speech production and perception. Pedagogically, however, until quite recently, this technology was not routinely employed as a learning-aid. Over the first half of the twentieth century, undergraduate phonetics courses delivered through lectures and (costly) face-to-face small group teaching began to be established. Gradually, too, hardware entered the learning environment—phonographs and tape-recorders provided sound recordings, kymographs, oscilloscopes, and spectrographs provided images of waveforms, formants and pitch contours, transforming teaching and learning of both theory and practical skills. Then, computers and the internet revolutionized education, including phonetics education. Audio materials were supplemented and enhanced with visuals and by the 21st century, interactive computer-assisted learning was well established. Today, technology can sometimes even replace the teacher completely. So now, in response to increasing numbers of financially driven course closures: can technology breathe new life into the subject of phonetics?

9:45–10:00 Panel Discussion

10:00–10:15 Break

Contributed Papers

10:15

3aSC6. Teaching phonetics in an active-learning classroom: The role of teaching assistants. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Active-learning classrooms (ALCs) facilitate innovative learner-centered pedagogical approaches. However, ensuring that all students receive appropriate support during learning activities can be challenging. Incorporating teaching assistants (TAs) is one avenue for increasing assistance available to students. Yet, investigations of the interactions among TAs and students in ALCs are lacking. To address this gap, a phonetics course with three TAs was studied. Three class sessions were video taped and students were given surveys to assess their perception of the learning activities and interactions with the TAs. Across sessions, 93% of students reported that their interactions with the TAs allowed them to move forward with the task. Similarly, 89% of students indicated that the TAs were able to guide them to the answer without giving them the answer. The video data suggest that the teaching assistants' interactions during small group activities differ from the instructors'. During the small group activities, the instructor spent 62% of the time in direct consultation with groups while the TAs only spent 30%. Additional analyses of the video and survey data will inform best practices for incorporating TAs into ALCs. [Work supported by a Scholarship of Teaching and Learning Grant from Indiana University.]

10:30

3aSC7. Teaching phonetics to undergraduate students majoring speech and hearing sciences and disorders. Makoto Kariyasu (Speech and Hearing, Kyoto Gakuen Univ., 18 Gotanda, Yamanouchi, Ukyou-ku, Kyoto 602-8577, Japan, kariyasu@kyotogakuen.ac.jp)

Phonetics is a fundamental course to provide students key knowledge and skills for understanding speech and hearing disorders. In our university, the students are required to take two courses. In Phonetics, we present the basic anatomy and physiology of speech production with emphasis on the three elements, flow, vibrations, and movement. Historical studies on speech production and perception, such as phonetic invariance, perceptual restoration, are presented. Based on the experimental findings, students learned the nature of human communication. That is, speech production is highly adaptive, but speech signal is somewhat varied. In some situations, the top-down processing helps listeners understand speech code. In Phonetics LAB, phonetic transcription, sound productions with vocal tract aerodynamics and acoustics are covered. The acoustic theory of speech production is presented, then the speech waveform and spectrogram of vowels and consonants are explained. Students are requested to show the understanding how speech is produced in reference to the vocal tract shaping and aerodynamics. For speech therapists, it is essential to demonstrate an automatic processing of hearing sounds and knowing the vocal tract. Disordered speech should be understood for possible mechanisms because speech therapists need to modify the human body and its control mechanisms.

10:45

3aSC8. Variability and invariants: Facilitating deep learning. Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover W225, Athens, OH 45701, leec1@ohio.edu)

Discussions of education in acoustics have traditionally focused on content issues such as "demonstration experiments, laboratory exercises, interpretation of basic concepts in acoustics, homework problems, possible exam questions, and tutorial papers" as noted in the announcement for a JASA special issue on education in 2010. However, well-constructed content does not always translate to actual learning. Furthermore, a common sentiment among new and established teachers is that much time and effort end up being spent on classroom management rather than teaching the content. How do students learn best? How do teachers help students understand the nature and progress of their learning? This paper reports a longitudinal study on teaching speech science to undergraduate students majoring in communication sciences and disorders. The course was taught by the author 19 times in a span of 10 years to 994 students at a public university. The relationships between how grades were determined, the actual final grade, and teaching evaluation are examined. The findings are discussed in the context of the author's reflections on teaching philosophy and strategies.

11:00

3aSC9. Flipping the phonetics classroom. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

The 21st century has seen a continued expansion of bandwidth, processor speed, and availability of multimedia recording tools for the average PC. Consequently, the flipped classroom model has gained popularity, in both K-12 and higher-level education. In a flipped classroom, the instructor typically records short lectures, to be viewed online by the students prior to class. Classroom time is then devoted to activities that might normally be assigned as homework, such as problem solving, examination of case studies, discussion or other interactive exercises. Advantages of the pre-recorded lecture include the opportunity for students to replay the lectures and the availability of captioning, while disadvantages of this format may include a lack of spontaneity or ability to stop and ask a question. The drawbacks may, however, be compensated for by the increased feedback opportunities for students, allowing teachers more opportunities to gauge gaps in student knowledge. The phonetics classroom seems particularly well suited for the increased opportunity for in-class exercises and feedback offered by the flipped classroom approach. This presentation will describe the author's implementation of the flipped classroom model in an undergraduate phonetics classroom, including a discussion of students' attitudes, technical issues, and other challenges.

11:15

3aSC10. Developing an online phonetics course for a diverse student population. Caroline L. Smith (Univ. of New Mexico, MSC 03 2130, Linguist, 1 University of New Mexico, Albuquerque, NM 87131-0001, caroline@unm.edu)

Approximately 140 students take Introduction to Phonetics each year at the University of New Mexico. They are diverse in several ways: in age (many are returning to school after years away), in ethnicity (UNM is majority-minority), in major (the course serves Linguistics and Speech & Hearing), and in preparation (there are no prerequisites, but many students have some related coursework). This paper reports on design considerations and experience with the first online offering of this course. Studying online requires self-motivation and the ability to work independently, which are difficult for the large number of students with poor study skills and time management. This course attempts to counter these challenges with a high degree of structure, and activities in a variety of formats for learning and assessment. Although some in distance education view structure as a "constraint" for learner and teacher (Farquhar 2013 *Eur. J. e-learning*), this instructor believes that the flexibility and isolation of online learning are overwhelming for many students and risk leading to paralysis. Providing options among different activities coupled with deadlines for completion of those they select, and requiring frequent interaction with classmates and instructor, is the chosen strategy for keeping students engaged and on track.

11:30

3aSC11. Online vs. in-person: Exploring the outcomes and impacts. Benjamin V. Tucker and Timothy Mills (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

Over the last few decades, the teaching of online courses has grown tremendously, especially with the rise in popularity of massive open online courses. In phonetics, the offerings of online courses has also grown. The University of Alberta has offered an online phonetics course for four years, which is academically equivalent to the in-person course. The University of Alberta online course runs as a standard semester course offered twice a year with the similar deadlines and structure as the in-person course. In this presentation we discuss and explore the advantages and challenges of online versus in-person courses from the instructor's perspective. We report the differences and similarities in student learning outcomes and student success. We discuss methods of student engagement, implementation of laboratory activities (active-learning), and exam invigilation. We also comment on the advantages of combining materials from online and in-person courses in blended learning environments.

11:45

3aSC12. Improving comprehension through team quizzes. Amelia E. Kimball, Patrick Drackley (Linguist, Univ. of Illinois at Urbana Champaign, 4080 Foreign Lang. Bldg. MC-168, 707 s Mathews Ave, Urbana, IL 61801, ameliak@bu.edu), and Jennifer Cole (Linguist, Northwestern Univ., Evanston, IL)

As instructors of a large introductory linguistics course, we face a challenge common to many teachers in the language sciences: we have a large amount of basic knowledge to communicate before moving on to more complex applications, and our students have varying degrees of preparation for our class. We address this challenge by using team-based learning (Michaelsen *et al*., 2001) and team quizzes (Gross Davis, 2009, p. 194). In this presentation, we share our approach to team work through a hands-on demonstration of the Immediate Feedback Assessment Technique, a scratch-off answer sheet that encourages students to work together to find the correct answer during a quiz. Using this technique, students move beyond passive listening as they explain concepts and defend their ideas to team members. We find that this technique works particularly well for introducing new topics that require repetition before "sticking." We also find that by keeping work focused in the classroom and providing peer feedback, students enjoy participating in team activities. Adopting this method has shown positive outcomes in our course, with use of team quizzes correlated with higher test scores, more positive course evaluations, and increases in course enrollment.

Session 3aSP

Signal Processing in Acoustics and Underwater Acoustics: Detection, Classification, Localization, and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) III

Ballard J. Blair, Cochair
Electronic Systems and Technology Division, MITRE Corporation, 202 Burlington Rd., Bedford, MA 01730

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

Invited Paper

9:00

3aSP1. Signal processing trade-offs for the quality assessment of acoustic color signatures. Brett E. Bissinger, J. D. Park (Appl. Res. Lab. at Penn State Univ., PO Box 30, State College, PA 16804, beb194@psu.edu), Daniel Cook (Georgia Tech Res. Inst., Smyrna, GA), and Alan J. Hunter (Univ. of Bath, Bath, United Kingdom)

Acoustic color is a representation of the spectral response over aspect, typically in 2-D. The two natural axes for this representation are frequency and the aspect to the object. It is intuitive to assume finer resolution in the two dimensions would lead to more information extractable for improved quality. However, with conventional linear track data collection methods, there is an inherent trade-off between signal processing decisions and the amount of information that can be utilized without loss of quality. In this work, how the information distribution is affected with choices of representation domain will be presented. The quality metrics including resolution, signal-to-noise-ratio, and other metrics will be discussed in the context of various signal processing choices and parameters. Other representation approaches as extensions of acoustic color will also be explored, such as time-evolving acoustic color that shows how the spectral response changes within a ping cycle.

Contributed Papers

9:20

3aSP2. Signature mining and analysis of urban environment data. Morris Fields and Hollis Bennett (Environ. Lab, USACE-ERDC, CEERD-EE-C, 3909 Halls Ferry Rd., Vicksburg, MS 39180, morris.p.fields@usace.army.mil)

When investigating low energy acoustic signatures such as infrasound, determining signal from noise can be complex, particularly in urban environments. Understanding the characteristics of all sources will assist in suppression of signatures deemed to be noise, and in the enhancement of the signatures of items of interest. The first part of the task is to determine the signatures of interest. These signatures will form the basis set for different classifiers. Development of classifiers for these signals of interest will involve categories such as stationary signals, transient signals, and types of infrastructure including utilities. Following classification, an idea of the usage can be determined based on signals from the infrastructure detected. The research to be presented focuses on signal extraction from a data set in order to begin development of classifiers and eventual detection algorithms for infrasound signals in urban environments.

9:35

3aSP3. Improving marine mammal classification using context from multiple hydrophones. Tyler A. Helble (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com), Regina A. Guazzo (Scripps Inst. of Oceanogr., UCSD, Chula Vista, CA), Stephen W. Martin (Marine Mammal Foundation, San Diego, CA), E. E. Henderson (SSC-PAC, San Diego, CA), Gabriela C. Alongi (Marine Mammal Foundation, San Diego, CA), Glenn Ierley (Scripps Inst. of Oceanogr., UCSD, Houghton, MI), and Cameron R. Martin (Marine Mammal Foundation, San Diego, CA)

Detection, classification, localization, and tracking (DCLT) of marine mammals is oftentimes performed in that order. However, in the sonar-signal processing communities and elsewhere, classification is usually the final step. Thus, more appropriately, the working order should be "DLTC." If classification is performed as the final step, the results can be greatly improved by using the *context* of the calls. By grouping likely calls into tracks, a collective of calls can provide much more information for classification than single calls alone. Additionally, when multiple species are calling at the same time, the location of the calls can be used to distinguish confusing signals. The time-series and spectral information of a call can also be enhanced by localizing first, and choosing the nearest hydrophone to the calling animal for signal analysis. If localization is not possible, classification can still be improved if two or more hydrophones are available with overlapping coverage, by using cross-correlograms. Multiple sensors also provide the ability to reduce detections due to sensor self-noise, and noise from fish and snapping shrimp. Collectively, these techniques were applied to vocalizing baleen whales on the Navy's Pacific Missile Range Facility,

and proved to greatly enhance the ability to classify Bryde's, humpback, fin, and minke whales.

9:50

3aSP4. Active learning for acoustic classification. Matthew G. Blevins, Edward T. Nykaza (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil), and William M. Nick (Comput. Sci., North Carolina A&T State Univ., Greensboro, NC)

Classification of noise sources based on their acoustic signatures is becoming increasingly necessary due to the increasing prevalence of noise monitoring systems. The intractable amount of data that these systems capture necessitates robust and efficient classification algorithms. However, the process of building a classifier requires manually labeling recorded noise events, which is a time consuming task. In this paper we explore a method for reducing the human burden and making the process more efficient called active learning, or "listener-in-the-loop." This method iteratively improves classifier performance by querying a human listener using optimally chosen observations based on certainty factors, which measure the degree of belief or disbelief based on probability for a particular classification. Classifier performance as well as algorithm efficiency, in terms of computational costs and amount of data required, will be discussed.

10:05–10:20 Break

10:20

3aSP5. Disambiguation of sonar target signatures using graph-based signal processing. Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Disambiguation and extraction of sonar target features in high-clutter and low signal-to-noise-ratio (SNR) environments have been well-known to be a daunting task. In particular, the elastic wave features evolve with time, aspect angle, sediment type, proud/buried status, among other factors. Furthermore, irregular geometry of buried and partially buried targets renders it challenging to separate target features unique to the target material against multiple reflections due to irregular target geometry, sediment effects, and other physical phenomena. This work attempts to apply the emerging field of graph-based signal processing to study associations between sonar target features unique to target material and target geometry. We will present our graph-based studies of acoustic color features over a complex field, i.e., magnitude and phase, with the objective of disentangling target features against interference introduced by the environment using association graph techniques. The talk will focus on features localized using both geometric and wavelet techniques, and present results on case studies across a variety of sediment types, aspect angles and proud and partially buried targets of different materials.

10:35

3aSP6. Chaos theory and topology for classification of marine mammals. Matthew Firneno, Kirk D. Bienvenu, Jack G. LeBien, Juliette W. Ioup, Nikolas Xiros, and Ralph Saxton (Physics, Univ. of New Orleans, 2000 Lakeshore Dr., 722 Huntlee Dr., New Orleans, LA 70148, mfirneno@uno.edu)

The Littoral Acoustic Demonstration Center–Gulf Ecological Monitoring and Modeling (LADC-GEMM) project collected underwater acoustic data in the northern Gulf of Mexico during the summer of 2015, with passive acoustic data recorded by the LADC-GEMM Environmental Acoustic Recording Systems (EARS). Such data have been widely studied in the past by Fourier/spectral analysis. Although highly useful, there are, however, novel perspectives that may be obtained with techniques from chaos theory and topology. Using a method known as time-delay embedding, a time-series manifold can be constructed from measured data. This manifold will be classified in terms of a topological score (measures of connectedness, compactness, etc.), in conjunction with its fractal and correlation dimensions. These combined features can potentially be used to produce a more robust classification means for the identification of marine mammals heard by the EARS buoys in the LADC-GEMM data. [This research was made possible in part by a grant from The Gulf of Mexico Research Initiative, and in part by an Internal Grant from the University of New Orleans Office of Research and Sponsored Programs. Data are publicly available through the Gulf of Mexico Research Initiative Information & Data Cooperative (GRIIDC) at https://data.gulfresearchinitiative.org.]

Session 3aUWa

Underwater Acoustics, Acoustical Oceanography, Physical Acoustics, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques III

David P. Knobles, Cochair
KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Invited Papers

7:40

3aUWa1. Nonlinear frequency dependence of sound attenuation in sea bottoms at low frequencies. Jixun Zhou (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332-0405, jixun.zhou@me.gatech.edu) and Zhenglin Li (State Key Lab. of Acoust., CAS Inst. of Acoust., Beijing, China)

In the 1980s, several papers were published that stressed the importance of non-linear frequency dependence (NLFD) of sound attenuation in marine sediments al low frequencies, and showed experimental evidence that the NLFD can have a significant effect on long-range sound propagation [Zhou *et al.*, *J. Acoust. Soc. Am.*, **78**, 1003–1009 (1985); **79**, Suppl. 1, S68, (1986); **82**, 287–292(1987); **82**, 2068–2074 (1987)]. According to Kibblewhite [*JASA*, **86**, 718–738 (1989)], "the only program directly related to marine sediment attenuation and not identified by Hamilton is that of Zhou *et al*." This analysis of NLFD was extended later to cover 20 locations in different coastal zones around the world, resulting in an effective geoacoustic model for sandy to sand-silt-clay bottoms. [Zhou, Zhang and Knobles, *JASA*, **125**, 2847–2866 (2009)]. However, the attenuation-frequency relationship is still an open issue. There is a pressing need to have quality sound-speed and attenuation data in the low- to high-frequency transition band from one location by using both inverse and direct techniques. Thus, a sea-going experiment in an area with shallower water depth is desirable. This paper will discuss some physical and technical issues related to broadband geoacoustic inversions and examine some data from muddy bottoms.

8:00

3aUWa2. Sediment sound speed inversion at low frequencies in shallow water. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

A direct method previously developed for shallow water inversion is presented here along with the assumptions for its convergence and improvements for the solution of a more complex inverse problem. The method estimates sediment sound speed and follows discontinuities as the sediment layer properties change. Sediment thickness is also calculated. The approach is fast in contrast to other techniques that navigate a multi-dimensional search space. Implementation requires low-frequency measurements at a horizontal array and a source anywhere in the water column with only the continuous spectrum available. Several assumptions are made which appear restrictive at first but can be eventually relaxed. Although the approach was first developed considering a constant density profile, we show how realistic density profiles can be considered. [Work supported by ONR.]

8:20

3aUWa3. An experimental benchmark for geoacoustic inversion methods. N. Ross Chapman (School Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca)

Over the past 25 years, many methods have been developed for estimating parameters of geoacoustic models of the ocean bottom from acoustic field data. The methods involved model-based comparisons between experimental data and calculated fields or field quantities. Benchmarking workshops based on simulated field data confirmed the ability of the methods for inverting geoacoustic model parameters for range-independent and range-dependent ocean environments. Although there are many published examples of geoacoustic inversions in various different ocean environments, the methods have not been tested in an experimental benchmark. This paper describes the design of an experimental benchmark for geoacoustic inversion methods, and presents an analysis of the performance of the methods with experimental data. The benchmark involves application of different inversion methods to data spanning a frequency band from 50to 2500 Hz that were obtained at a single well-surveyed ocean bottom site. The performance metric involves comparisons of: estimated geoacoustic profiles against the ground truth information of the site; modal wavenumbers for the inverted models; and transmission loss calculations based on the estimated models against measured data. The analysis shows that the methods are capable of estimating realistic geoacoustic profiles, with sound speed the most reliable estimated model parameter.

8:40

3aUWa4. Sediment characterization through normal-incidence echo sounding at the SeaBed Characterization Experiment 2017.
Michael Rukavina and Marcia J. Isakson (Univ. of Texas at Austin: Appl. Res. Labs., 10000 Burnet Rd., Austin, TX 78758, michael.rukavina@utexas.edu)

The Seabed Characterization Experiment (SBC17) was conducted in February 2017 off the east coast of the United States in the New England Mudpatch. The overall mission of the experiment was to determine the effect of the seabed on acoustic propagation. However, it was determined from pre-surveys that the area also contained a significant concentration of benthic biology. In order to characterize the variability of the sediment and the influence of the biology, the Applied Research Laboratories at the University of Texas (ARL:UT), deployed an acoustic system to collect normal-incidence echo returns from a 6 to 20 kHz chirp. The system was deployed in various locations around the mud patch to obtain a diverse set of data. The data displayed a long coda, which has been previously linked to the presence of benthic biology. The ability of these data to predict and measure sediment type and benthic biology concentration will be assessed. [Work sponsored by ONR, Ocean Acoustics.]

Contributed Papers

9:00

3aUWa5. An application of Bayesian inference techniques to compare three competing sandy sediment models. Anthony L. Bonomo and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Many competing geoacoustic models exist to represent sandy sediments. In this work, Bayesian inference is employed to compare the sensitivity of the physical input parameters of three competing models: the viscous grain shearing model of Buckingham, the Biot-Stoll model, and the extended Biot model of Chotiros, using previously taken wave speed and attenuation laboratory measurements. Bayesian model selection techniques are then employed to assess the degree to which the predictions of these models can be used to explain the observed data. Since all three geoacoustic models require a large number of input parameters and many of these input parameters are difficult to measure, a main goal of this work is to use Bayesian techniques to determine which parameters can be resolved via inversion and what kinds of data are required for parameter resolution. [Work supported by ONR, Ocean Acoustics.]

9:15

3aUWa6. Inversion of geoacoustic parameters from transmission loss measurements in the presence of swim bladder bearing fish in the Santa Barbara Channel. Orest Diachok (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com) and Altan Turgut (Naval Res. Lab., Washington, DC)

It is well established that attenuation due to swim bladder bearing fish can have profound effects on transmission loss (TL) in continental shelf and slope environments (Weston, 1967; Diachok, 1999). Realistic estimates of geoacoustic parameters from TL measurements in such environments may be derived by application of the concurrent bio and geo inversion method (Diachok and Wales, 2005; Diachok and Wadsworth, 2014). This method assumes that the biological environment may be characterized by bio-alpha (attenuation coefficient within the layer), layer depth, and layer thickness. Application of this method to broadband TL measurements between 0.3 and 5 kHz in the Santa Barbara Channel resulted in realistic estimates of geo and bioacoustic parameters. Inverted geoacoustic parameters were consistent with coincident core and chirp sonar measurements. Inverted bioacoustic parameters were consistent with concurrent measurements of fish depths and length distributions of dominant species. The latter permitted calculation of swim bladder dimensions and resonance frequencies. Neglect of bio-alpha resulted in unrealistically high values of geo-alpha. These results suggest that some previously reported inversions of geo-alpha from TL measurements at other sites may have been biased by neglect of bio-alpha. [This research was supported by the ONR Ocean Acoustics Program.]

9:30

3aUWa7. Inversion of geoacoustic parameters from transmission loss measurements in the presence of swim bladder bearing fish in the Yellow Sea. Orest Diachok (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

Realistic estimates of geoacoustic parameters from transmission loss (TL) measurements in the presence of swim bladder bearing fish may be derived by application of the concurrent bio and geo inversion method (Diachok and Wales, 2005; Diachok and Wadsworth, 2014). This method was applied to Qiu's, (1999) TL measurements in the Yellow Sea. Qiu's measurements were made in August, south of the Shandong Peninsula, where the bottom is composed of sand, and the concentration of anchovies, the dominant species in the Yellow Sea, is high, at night when anchovies are dispersed near the surface and bio-alpha is relatively high. The measured attenuation coefficient (TS—cylindrical spreading—absorption) was nearly 8 dB/km at the resonance frequency of anchovies. Inverted values of bioacoustic parameters were consistent with bioacoustic parameters of anchovies. Inverted geoacoustic parameters were consistent with geoacoustic properties of sand. Disregard of the effects of bio-alpha on TL resulted in unrealistically high estimates of geo-alpha at frequencies above a few hundred Hz. These observations suggest that some previously reported inferences of geo-alpha from TL measurements in the Yellow Sea may have been biased by neglect of bio-alpha. [This research was supported by the ONR Ocean Acoustics Program.]

9:45

3aUWa8. The development and experimental study of the Ballast *in situ* Sediment Acoustic Measurement System. Guanbao Li (First Inst. of Oceanogr., State Oceanic Administration, Qingdao, China), Baohua Liu (National Deep Sea Ctr., SOA, Qingdao, Shandong, China), Guangming Kan (First Inst. of Oceanogr., State Oceanic Administration, Qinfdao, Shandong, China), Jingqiang Wang, and Xiangmei Meng (First Inst. of Oceanogr., State Oceanic Administration, No. 6 Xianxialing Rd., Laoshan District, Qingdao, Shandong 266061, China, wangjqfio@fio.org.cn)

A ballast *in situ* sediment acoustic measurement system (BISAMS) was newly developed. The mechanical structure, the function modules, the working principles, and a sea trial will be reported in this study. The system relies on its own weight to insert transducers into seafloor sediments and can accurately measure the penetration depth using a specially designed mechanism. The system comprises an underwater position monitoring and working status judgment module and has two operation modes: self-contained measurement and real-time visualization. The designed maximum working water depth of the system is 3000 m, and the maximum measured depth of seafloor sediment is 0.8 m. The system has 1 transmitting transducers and 3 receiving transducers. The transmitting frequency band is 20–120 kHz. The *in situ* acoustic measurement system was tested at 15 stations in the northern South China Sea, and the repeated measurements in seawater demonstrated good working performance. Comparison with predictions from empirical equations indicated that the measured speed of sound and attenuation matched with the predicted values and that the *in situ* measured data were reliable.

10:00

3aUWa9. In situ and laboratory acoustic attenuation measured in the sediments of the southern Yellow Sea. Xiangmei Meng (First Inst. of Oceanogr., No. 6 Xianxialing Rd., Qingdao 266061, China, 780610@sina.com), Jingqiang Wang, Qingfeng Hua, Guanbao Li, and Guangming Kan (First Inst. of Oceanogr., Qingdao, Shandong, China)

Acoustic attenuation directly determines sound propagation distance in seafloor sediments. Moreover, by studying the attenuation law of sound propagation in sediments, much information about sediment properties can be obtained. In June 2009 and June 2010, *in situ* measurements of acoustic attenuation at 30 kHz were made in the sediments of the southern Yellow Sea. Meanwhile, sediment cores were collected and laboratory measurements of acoustic attenuation between 25 and 250 kHz and physical properties were conducted. We compared *in situ* and laboratory attenuation measured in the sediments and analyzed the differences. The frequency dependence of acoustic attenuation of silt, silty clay, and clay is discussed on the basis of laboratory measurements. Combining with sediment physical properties, we analyzed the acoustic attenuation mechanism of silt, silty clay, and clay.

10:15

3aUWa10. Gas-bubble inversion in marine sediments. Guangying Zheng, Yiwang Huang, and Jian Hua (Harbin Eng. Univ., Nantong St. No. 145, Harbin 150001, China, 276454158@qq.com)

The acoustic properties of marine sediments can dramatically change because of the presence of gas-bubbles. Many applications require the detailed information of gas-bubbles, such as gas void fraction and gas-bubble size distribution. It is possible to relate the acoustic transmission measurements with bubble sizes. This study aims toward the development of an acoustic method able to both detect and quantify the gas present in marine sediments. This acoustic method adapts the effective density fluid model corrected by gas-bubble pulsations as a forward model and expands the unknown gas-bubble size distribution by a finite sum of cubit B-splines. The inverse problem can be transformed into solving the equation groups involving the coefficients of cubit B-splines. This method can be verified by testing analytical results and then applied to measurement sound speed and attenuation data which were acquired via transmission experiments.

10:30

3aUWa11. Laboratory measurements of sound speed and attenuation dispersion in calcareous sediments from coral reefs. Jingqiang Wang (First Inst. of Oceanogr., State Oceanic Administration, No. 6 Xianxialing Rd., Laoshan District, Qingdao, Shandong 266061, China, wangjqfio@fio.org.cn), Baohua Liu (National Deep Sea Ctr., SOA, Qingdao, Shandong, China), Guangming Kan, Guanbao Li, and Xiangmei Meng (First Inst. of Oceanogr., State Oceanic Administration, Qingdao, Shandong, China)

The calcareous sediment is an important type seafloor sediment exist in coral reef island sea areas. The acoustic properties of calcareous sediment are significant for modeling sound propagation and underwater reverberation. In order to analyze the frequency dependence of sound speed and attenuation in calcareous sediments, the sediments were firstly screened and remodeled in plexiglass tubes in laboratory, and the sound speed and attenuation were measured at the frequency range of 27–247 kHz. The grain sizes of sediment samples were $<$0.075 mm, 0.075-0.5 mm, 0.5-1 mm, 1–2 mm, and 2–4 mm, respectively. The sound speeds of different grain-size sample were 1564.83–1607.36 m/s, 1564.82–1607.36 m/s, 1527.76–1553.69 m/s, 1529.40–1594.72 m/s, and 1541.87–1596.51 m/s, respectively. The attenuation were 24.94–206.35 dB/m, 20.78–208.75 dB/m, 9.66–271.94 dB/m, 12.33–310.36 dB/m, and 12.60–293.60 dB/m, respectively. The sound speeds and attenuation were found to increase remarkable with frequency. The fine grained sediments have higher sound speed than the coarser sediments, which due to the higher bulk density and lower porosity of fine grained sediments. The dispersion gradients of sound speed and attenuation in coarser sediments were more remarkable than that in fine-grained sediments, which may due to the higher tortuosity of coarser sediments.

3a WED. AM

Session 3aUWb

Underwater Acoustics: Session in Honor of Chester McKinney

Thomas G. Muir, Cochair
Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Clark Penrod, Cochair
Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Chair's Introduction—8:00

Invited Papers

8:05

3aUWb1. Chester McKinney, champion in acoustics. David T. Blackstock (Appl. Res. Labs & Dept. of Mech. Eng., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

Chester M. McKinney (1920–2017) was born and grew up in Cooper, Texas. Having built radios as a boy, he majored in physics at East Texas State Teachers College, taught high school science back in Cooper, and joined the U.S. Army immediately after Pearl Harbor. As a radar officer in the Army Air Corps, he did course work at Harvard and MIT in radar and electronics, and flew many missions in B-29s over China, India, and the Pacific. After the war he earned M.S. (1947) and Ph.D. (1950) degrees in physics at University of Texas, working primarily in electromagnetics at the University's Defense Research Laboratory (DRL, later renamed Applied Research Laboratories, ARL). In 1948 he married Linda Hooten. After a short stint at Texas Tech University, he returned to DRL in 1953 and switched from electromagnetics to acoustics, primarily underwater sound, which became his life's work. After 15 years as ARL Director, Chester retired in 1980. Having joined the Acoustical Society of America (ASA) in 1953, he chaired the Austin Meeting in 1975. He and Linda were fixtures at ASA Meetings. He became Vice President in 1984–1985 and President in 1987–1988. ASA awarded Chester its Gold Medal in 2005.

8:25

3aUWb2. Chester McKinney and the Acoustical Society of America: Over sixty years of service. Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

The impact of Chester McKinney on the Acoustical Society of America cannot be overstated. Chester served as a charter member of the Underwater Acoustics Technical Committee and was its first chair in 1956. He was the driving influence in making the ASA the professional home for underwater acousticians. He established three committees that still serve the society: Archives, Tutorials and Public Relations. He served on the Executive Council, the Medals and Awards Committee, the Committee on Meetings and the Nominating Committee among others. Chester went on to serve as vice-president and president of the society. In 1989, Chester pioneered a census of acousticians. He chaired the census ad-hoc committee, collected and organized the responses and reported the results in JASA [McKinney, Hurdle, and Blue, "A profile of the acoustics community in the United States and Canada," *J. Acous. Soc. Am.*, **91**(2):1169–1179, 1992]. This census can now be compared with the statistics of today to track the growth of the society. Chester received the Gold Medal in 2004, the highest honor the society can bestow.

8:45

3aUWb3. Chester McKinney's legacy at Applied Research Laboratories, The University of Texas. Clark Penrod (Appl. Res. Labs, The Univ. of Texas, PO Box 8029, Austin, TX 78713-8029, penrod@arlut.utexas.edu)

Chester McKinney joined the ARL:UT staff as our first graduate student employee in 1946. Subsequently, he was the prime mover in establishing ARL's high frequency acoustics program, in which he retained a strong interest throughout career. However, Chester's contributions were by no means confined to research in underwater acoustics. Chester served as ARL's second director from 1965-80, and it was certainly the case that his efforts to establish ARL as a great laboratory were as important as anything he did. His thoughts and philosophy about how contract laboratories should operate in a university setting continue to guide ARL today. Chester believed in ARL having strong involvement in scientific research, that staff should participate in scholarly societies such as ASA and publish their work in archival journals, and that we should engage students in research. Alongside these activities, he also believed we should have strong engagement in applications, system development, and prototyping. He felt that the academic and applied areas of emphasis would complement each other, and lead to a more productive lab. ARL's current status as a healthy university laboratory owes much to our continued adherence to the set of guidelines that Chester formulated five decades ago.

9:05

3aUWb4. Chester McKinney: Lessons I learned from a prominent scientist and leader in our acoustics community. William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

The impression that was left with me from his speech after he received the ASA's Gold Medal at the 1994 New York City meeting still remains indelible. It was exactly the way someone born in Cooper Texas, population 2,600 should communicate the excitement of his career journey to national and international leadership in science, academia, and the technology associated with national defense. Comfortable in front of a large audience, he walked back and forth on the stage with complete informality reminiscing about his experiences in the multiple worlds he inhabited. As the new director of Marine Physical Lab, SIO/UCSD one generation after his tenure as director of the Applied Research Labs, UT, he reaffirmed to me why he had served as my mentor by example. Here, I review some of the lessons that his career provided

9:25

3aUWb5. Chester McKinney: A legacy of high frequency sonar development at Applied Research Laboratories, The University of Texas at Austin. John Huckabay (Appl. Res. Labs., The Univ. of Texas at Austin, PO Box 8029, Austin, TX 78759, huckabay@utexas.edu)

Following the Korean War, improvements in high frequency, high resolution sonar were needed in order to successfully prosecute naval mines. Chester McKinney led an effort at Applied Research Laboratories (formerly Defense Research Laboratories), The University of Texas at Austin (ARL:UT), to make progress in this important area of research; and began a lifelong passion for Chester to pursue both basic and applied research in high frequency acoustics as applied to mine countermeasures. This research has been performed over many years by numerous scientists and engineers at ARL:UT and continues today. Based on this research, both experimental and operational high frequency, high resolution sonars were constructed, tested, and employed. This paper will explore some of these important sonar developments as well as some of the underlying physics. These developments span from analog implementations of CTFM sonars, pulse sonars, beamformers, and signal processing to early implementations of digital sonars and digital signal processing. Key parts of sonar research will be traced from the AN/UQS-1 to the AN/SQQ-14, AN/SQQ-16, AN/WQS-1, and the Data Collection Test Bed Sonar (development model for the AN/SQQ-32).

9:45

3aUWb6. The McKinney-Anderson paper and its impact on sediment acoustics. Nicholas P. Chotiros (U.S. Office of Naval Res. Global, London, United Kingdom) and Anthony L. Bonomo (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The earliest comprehensive, *in-situ* measurements of the acoustic backscattering strength of the seabed as a function of frequency, grazing angle and sediment type were made by McKinney and Anderson, and published in the society's journal in 1964. It was a landmark paper and laid the groundwork for subsequent measurements and models. It is widely referenced in books and papers where seabed scattering and sonar performance prediction are concerned. Scattering strength and sediment classification, usually quantified in terms of the mean grain diameter, were thought to be closely connected. In the laboratory, using pristine samples of sorted sand, this connection was clearly demonstrated, but *in situ* it was overwhelmed by other factors, such as uneven size distributions, biological activity and roughness. The underlying nature of the seabed is also reflected in the scattering strength. For the same roughness, density and sound speed, the solid/fluid model consistently overestimates the scattering strength because it cannot account for the relative motion between grains and pore water. Thus, recent advances in poroelastic modeling can shed new light on the measurements of McKinney and Anderson. [Work supported by ONR, Ocean Acoustics Program.]

10:05–10:20 Break

10:20

3aUWb7. Acoustical research and publications of Chester McKinney. Thomas G. Muir (Appl. Res. Laboratories, Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Chester McKinney and his students and colleagues began publishing on acoustics research in the early 1950s, in *The Journal of the Acoustical Society of America*, *The Journal of Underwater Acoustics*, and in numerous proceedings of symposia dealing with high resolution sonar for mine hunting. Much of his work involved official studies on major issues, done for and published by the U.S. Navy. A number of his lectures to lay audiences addressed timely topics pertaining to contemporary topics, such as the conduct of classified research on university campuses. He was noted for excellent tutorials and fascinating historical perspectives. Some of Chester's publications are briefly discussed here, including his *Journal* publications on transduction with reflectors, the transition between near and far-field radiation, the target strength of geometric objects and the discovery of creeping or circumferential waves in echoes. [Work supported by Applied Research Laboratories, the University of Texas at Austin.]

10:40

3aUWb8. McKinney Fellowship in acoustics at ARL:UT. Kyle S. Spratt and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, sprattkyle@gmail.com)

The McKinney Fellowship in Acoustics is a graduate research fellowship awarded to one student per year by Applied Research Laboratories at The University of Texas at Austin (ARL:UT). Created in 2006 and named after former lab director Chester M. McKinney, the fellowship is meant to foster quality research in acoustics while maintaining the strong connection between ARL:UT and the Graduate Program in Acoustics in the Cockrell School of Engineering. The fellowship provides up to three years of tuition and fees as well as a graduate research assistantship and travel expenses for the purpose of performing work toward a master's or doctoral degree on a topic related to the acoustics research and development performed at ARL:UT. In this talk, the implementation of the McKinney Fellowship program will be given, as well as an overview of the research that has been done by the 10 McKinney Fellows of the past decade. The first author, a past McKinney Fellow, will also offer some personal perspective on the program.

WEDNESDAY AFTERNOON, 6 DECEMBER 2017 STUDIO 9, 1:15 P.M. TO 3:00 P.M.

Session 3pAA

Architectural Acoustics: Speech Privacy Concerns in Open Plan Spaces

Kenneth W. Good, Chair
Armstrong, 2500 Columbia Ave., Lancaster, PA 17601

Chair's Introduction—1:15

Invited Papers

1:20

3pAA1. Balancing the detrimental effects of office noise annoyance and distraction on work performance. Martin S. Lawless, Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

Broadband, steady-state background noise can improve open office conditions by facilitating speech privacy and reducing distraction caused by intermittent, occupancy-generated noise. The background noise is typically generated by HVAC systems, though can be added with loudspeakers to boost speech masking. However, too high background noise levels can cause annoyance, fatigue, and other noise-related symptoms. It is yet unclear whether noise annoyance or distraction impairs work performance more. This study investigated the trade-off between noise annoyance and distraction, as well as their effects on acoustic dissatisfaction and performance. Subjects performed cognitive tasks while exposed to simulated office acoustic environments reproduced using higher-order Ambisonics. At fixed time intervals, the subjects could change the acoustic environment by adjusting either the background or intermittent noise levels. Lowering background noise caused the intermittent noise to rise, and vice versa. By the end of testing, it was expected that each subject equalized their dissatisfactions of the two noise types. Annoyance and distraction were assessed with a survey at each time interval. Physiological measures, including heart rate variability and skin conductance, were collected to correlate arousal/stress levels with each acoustic environment. The results of the study may provide context to effectively utilize background noise in open-plan offices.

1:40

3pAA2. Using ODEON acoustical modeling software to predict speech privacy in open-plan offices: Part 2. Valerie Smith, Jason R. Duty, and Diego Hernandez (Salter Assoc., 130 Sutter St., Fl. 5, San Francisco, CA 94104, jason.duty@cmsalter.com)

Speech Privacy Index is one of the commonly used metrics to discuss an occupant's acoustical comfort in an open-plan office. This paper continues our discussion from the Boston '17 conference on using the ODEON acoustical modeling software to predict the speech privacy index in open-plan offices. Our original paper examined the effects of using the ODEON provided equation for Articulation Index to calculate Privacy Index, compared these results to "real world" measurements, and attempted to adjust this equation to better fit the Speech Privacy Index (per ASTM 1130-16). In this paper, we will update our analysis to include a larger open-place office floorplate. To continue the discussion, we will also analyze the variation in results, if any, due to changing the scattering coefficients for various materials to determine if this significantly affects the modeled speech privacy results.

2:00

3pAA3. Possible path for speech privacy design and performance approaches. Kenneth P. Roy (Armstrong, 2500 Columbia Ave, Lancaster, PA 17603, kproy@armstrongceilings.com)

Speech privacy (or the lack thereof) is becoming a well-defined issue relating to "acoustic comfort" as a factor in building IEQ. This is the case for commercial offices as well as for healthcare spaces. The drivers for this include worker productivity, and confidentiality of sensitive information. CBE (Center for the Built Environment, UC Berkeley) surveys show the lack of speech privacy as being the most significant factor in disapproval of building IEQ for "green rated," and for all other building types. Healthcare regulations such as HIPAA and HCAHPS are transforming the design of hospital facilities and procedures in part due to oral privacy and annoyance issues. The "WELL Building Standard" is going beyond LEED to consider the occupants health and productivity as affected by the IEQ.

ASHRAE is going beyond "high performance green buildings" to now include an IEQ Global Alliance. So, where are we going with all this work?? Well, the metrics and measurement procedures used to measure and define levels of speech privacy are being updated at ASTM International to provide a basis for evaluation of speech privacy in both open plan and closed plan, and combinations thereof. This work will be outlined herein.

2:20

3pAA4. Evaluation of ambient noise including conversation in medical facilities. Yumi Koyama (School of Pharmacy, Nihon Univ., 7-7-1 Narashinodai, Funabashi, Chiba 274-8555, Japan, koyama.yumee@nihon-u.ac.jp) and Yasushi Shimizu (Sound/Form Design Lab., Hamamatsu, Japan)

We are studying the research focusing on an ambient noise in medical facilities with the aim of improving the acoustical environment, which can make patients comfortable to cure own illness at a medical area such as a patient room. At medical areas, there are many sound sources such as device sounds, personal active sounds, and talker's voices during a conversation. It is necessary to develop a method which can record and analyze an ambient noise without any intelligible contents of conversation voices, since all personal information belonging to a patient should be protected as confidential in medical facilities. In this experiment, acoustical measurement of each environmental sound such as the sound of footsteps, equipment alarm, intravenous stand, and medical cart at the simulated patient room in my school was conducted. Then, recorded ambient noises mixed with conversation voices and the other environmental sounds were fragmented in time, and the fragmented signal components were removed alternately so that the original conversation was unintelligible. The results are presented on the appropriate time duration of fragmentation and audio signal removal processing, and on the also prediction of each environmental sound except conversation voices for the processed audio signal.

2:40

3pAA5. Case study: Open plan "closed" offices or closed plan "open" offices. Kenneth W. Good (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

In today's built environment designers likes to blur the lines between categories and closed and open offices are no exception. Spaces are being built with high walls and doors but no ceilings. The occupants often call these "private" offices but how "private" are they? Are they open spaces or closed spaces, what are the expectations and how should we handle these spaces? Past experiences in these hybrid spaces have led me to question our measurement techniques. This case study will explore back to back measurements using the closed office method of ASTM E 336 vs the open plan method of ASTM E 1130 each calculated for Privacy Index (PI) in the same space and conditions.

WEDNESDAY AFTERNOON, 6 DECEMBER 2017 BALCONY M, 1:00 P.M. TO 2:30 P.M.

Session 3pBA

Biomedical Acoustics: Imaging I

Kausik Sarkar, Chair

George Washington University, 801 22nd Street NW, Washington, DC 20052

Contributed Papers

1:00

3pBA1. Quasi-static acoustic tweezing for low-volume blood coagulation analysis. Daishen Luo (Dept. of Biomedical Eng., TulaneUniv., 6823 St. Charles Ave., 440 Lindy Boggs Bldg. New Orleans, LA 70118, dluo@tulane.edu), Ray Holt (Mech. Eng., Boston Univ., Boston, MA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Available contact assays for blood plasma and whole blood coagulation have low predictive power in patients with coagulopathy or take a significant amount of time and blood volume to obtain diagnostic data. We have developed an innovative low-volume non-contact technology for real-time assessment of blood coagulation, referred to as "Quasi-static Acoustic Tweezing Thromboelastometry"(QATT). In our method, human blood drops with volume less than 5 microliter (~100 times smaller than the volume required by current coagulation technologies) are levitated in air by acoustic radiation forces. The sample drop location and deformation are induced by a quasi-static change in the acoustic pressure. By extracting a linear regime slope, the samples exhibit a unique elasticity profile over time (tweezograph) less than 20 minutes, characterized by clot initiation time (CIT), time to firm clot formation (TFCF), and maximum clot strength (MCS). The exposure of blood samples to pro- or anti-thrombotic agents (Fibrinogen and GPRP, respectively) led to significant changes in tweezographs within 10 minutes, thus allowing detection of hyper- or hypo-coagulable states. The advantages of small sample size, non-contact and rapid measurement make this technique desirable for real-time monitoring of blood coagulation in neonatal and pediatric patients with coagulation abnormalities.

1:15

3pBA2. Acoustic and atomic force microscopy characterization of microbubbles with varying shell chemistry. Mitra Aliabouzar (Mech. Eng., George Washington Univ., Mech. and Aerosp. Eng. 801 22nd St., Washington, DC 20052, mitraaali@gwmail.gwu.edu), Babak Eslami (Dept. of Mech. Eng., Univ. of Maryland, College Park, MD), Krishna N. Kumar, Santiago Solares, and Kausik Sarkar (Mech. Eng., George Washington Univ., Washington, DC, DC)

Applications of microbubbles (MBs) in diagnostic and therapeutic interventions critically depend on their stability and scattering properties. The shell chemistry of MBs defines these properties. We investigated the effects of shell chemistry on the size, abundance, acoustic response, and mechanical properties of MBs by varying the poly(ethylene glycol) (PEG) molar ratio (0 to 100%) in a two-lipid (DPPC and DPPE-PEG2000) component shell formulation. Increasing PEG concentration from 0% to 10% resulted in an increase in the number of MBs by at least 10-fold, with adverse effects upon further increases. Microbubbles made with 5–10% PEG generated the strongest fundamental as well as nonlinear (subharmonic and second harmonic) components at the excitation frequency of 2.25 MHz. We used interfacial rheological models to determine the mechanical properties of MB shells as functions of PEG concentration using experimentally measured attenuation values. We also employed atomic force microscopy (AFM) to perform thin planar film characterization of the shells. The correlation between the AFM measurements of film properties and the acoustic responses of the corresponding coated MBs will be discussed.

1:30

3pBA3. Acoustic characterization of 3D printed micro-structured scaffolds for tissue engineering. MITRA ALIABOUZAR (Mech. Eng., George Washington Univ., Mech. and Aerosp. Eng. 801 22nd St., Washington, DC 20052, mitraaali@gwmail.gwu.edu), Lijie Grace Zhang, and Kausik Sarkar (Mech. Eng., George Washington Univ., Washington, DC)

The acoustic and mechanical properties of 3D-printed porous poly-(ethylene glycol)-diacrylate (PEGDA) hydrogel scaffolds, as a widely used biomaterial, with different geometric channels (hexagonal and square) were explored using a pulse echo technique. The measured values of attenuation and speed of sound were found to be within the range of reported values for soft tissues making PEGDA scaffolds a suitable candidate for cartilage tissue engineering. We also showed that these properties as well as Young's modulus can be controlled and adjusted to desired values close to biological tissues by varying the 3D printing parameters. Furthermore, our 5-day proliferation as well as three-week chondrogenic differentiation results revealed that cell growth and tissue formation depend on the geometrical features of the 3D-printed scaffolds as well. Cell adhesion and proliferation greatly improved for scaffolds with square and hexagonal pore geometries compared to nonporous scaffolds. Scaffolds with square pores were determined to be the optimal for hMSC growth as well as chondrogenic differentiation. In addition to cell number, total collagen and Glycosaminoglycan (GAG) synthesis were 35% and 31% higher, respectively, for the square pattern compared to the hydrogel scaffolds with hexagonal pore shapes.

1:45

3pBA4. Ultrasound and lipid-coated microbubbles for osteogenic differentiation of mesenchymal stem cells in 3D printed tissue scaffolds. Jenna Osborne (George Washington Univ., Washington, DC), Mitra Aliabouzar (George Washington Univ., Laurel, MD), Xuan Zhou, Raj Rao, Lijie G. Zhang, and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

With increasing incidence of bone disorders in a rapidly aging, sedentary and overweight population, engineered bone tissues promise to be a better alternative for conventional bone grafts. However, bone tissue engineering currently suffers due to the inability to form mechanically strong porous structures that can be grown quickly. In this study, lipid-coated microbubbles (MB), traditionally used for contrast enhanced ultrasound imaging, were applied to harness the beneficial effects of ultrasound stimulation on proliferation and osteogenic differentiation of human mesenchymal stem cells (hMSCs) on 3D printed poly(lactic acid) (PLA) scaffolds. A significant increase in cell number was observed with low intensity pulsed ultrasound (LIPUS) treatment in the presence of MB after 1, 3, and 5 days of culture on scaffolds. Total protein content, alkaline phosphatase activity, and total calcium content were also found to increase with LIPUS with and without MB indicating enhancement in osteogenic differentiation. Integrating LIPUS and MB appears to be a promising strategy for bone tissue engineering and regeneration therapies.

2:00

3pBA5. Mechanism of echogenicity of echogenic liposomes. Krishna N. Kumar (Dept. of Mech. Eng., The George Washington Univ., 800 22nd St. NW, Washington, DC 20052, krishnagwu@gwu.edu), Sanku Mallik (Dept. of Pharmaceutical Sci., North Dakota State Univ., Fargo, North Dakota, Fargo, ND), and Kausik Sarkar (Dept. of Mech. Eng., The George Washington Univ., Washington, DC)

Liposomes prepared by a freeze-drying technique in the presence of mannitol have proved to be echogenic. However, the mechanism of echogenicity is not well understood. Here, we attempt to explain it. It was observed that only freeze-dried mannitol (without lipids) generates a strong scattered response because it generates bubble upon dissolution in water. The bubble generation was confirmed optically under an optical microscope. During the dissolution of the crystalline mannitol, the concentration of mannitol becomes locally very high. As the solute (mannitol) concentration increases, the saturated dissolved gas concentration decreases. Therefore, the dissolved gas in the solution near the dissolving crystal is in a supersaturated state. Upon sufficient supersaturation, bubble nucleation takes place. We found that freeze-dried crystalline excipients such as mannitol facilitate bubble nucleation compared to freeze-dried glassy excipient such as trehalose because of differences in surface morphology.

2:15

3pBA6. Simulation of multi-source external mechanical vibration for shear wave elastography. Heng Yang, Alex Benjamin, and Brian Anthony (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Bldg. 35, Rm. 231, Cambridge, MA 02139, hankyang@mit.edu)

Shear wave elastography (SWE) provides clinical diagnostics by probing the mechanical properties of soft tissue. Commercial ultrasound scanners use focused acoustic radiation force (ARF) to generate the requisite shear waves. This technique requires complex front-end electronics and may induce thermal stresses within the tissue and electronics. External mechanical vibration (EMV) based SWE emerges as a viable alternative; in addition to being low-cost and robust, it potentially offers higher local displacements and increased standardization compared to ARF based SWE. We present the simulation and sensitivity analysis of a novel EMV technique that can be adopted in clinical settings. The proposed design involves the use of two solid spheres placed symmetrically across or within the imaging plane of a commercial probe. These spheres can vibrate over a range of frequencies, amplitudes, and phases that are independent of one another. A non-linear, viscoelastic, finite-element model is used to simulate the propagation of induced shear waves within tissue-mimicking domains. A systematic sensitivity analysis is performed to determine the dependence of the amplitude, symmetry, attenuation, and interference of the induced shear waves on the following: spatial placement of the spheres relative to the ROI, distance between the spheres, their individual sizes, frequencies, amplitudes, and relative phases of vibration.

Session 3pIDa

Interdisciplinary: Hot Topics: Hunt Is Still Hot

Christina J. Naify, Chair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr., Pasadena, CA 91109

Chair's Introduction—1:00

Invited Papers

1:05

3pIDa1. Hydronephones: Acoustic receivers on unmanned underwater vehicles. Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu)

Noun | hy-drone-phone | \hi-drōn-fōn\ : An Instrument for listening to sound, mounted on an unmanned underwater vehicle. Yes, I made that word up—you heard it here first! The acronyms "UUV" (Unmanned Underwater Vehicle) and "AUV" (Autonomous Underwater Vehicle) have been buzzing around the underwater community recently, along with the term "underwater drone," which gained popularity in the media due to an international incident in December 2016 regarding an ocean glider (a subclass of AUV). Unmanned underwater vehicles are quickly becoming ubiquitous in the world of oceanographic sensing, military operations, and oil and gas exploration. Because sound travels far and fast underwater, it is an enabling mechanism for navigation and communication for these vehicles. "Hydronephones," specifically ocean gliders, are being used by our underwater acoustics community as receivers for long-range acoustics and marine mammal sensing. Key advantages of these vehicles are that they can be deployed for long durations of time in harsh environments and that they are much more cost-effective than traditional ship-based observational methods. An overview of this exciting technology will be presented along with details of some recent and ongoing "hydronephone" projects in underwater acoustics.

1:25

3pIDa2. Virtual reality meets architectural acoustics. Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

Computer programs for simulation and auralization of sound fields in rooms and of sound transmission in buildings became standard tools for architectural design and consulting. Another field of rapid progress is Virtual Reality—the combination of real-time signal processing with user interaction and multimodal human-machine interfaces. In this contribution, the development of simulation tools in architectural acoustics and further work aiming at real-time Acoustic Virtual Reality systems are reviewed and discussed with emphasis on the challenges and interdisciplinary solutions involving architectural acoustics as well as signal processing and psychoacoustics.

1:45

3pIDa3. Hospital noise: How bad is it? Ilene Busch-Vishniac (BeoGrin Consulting, 200 Westway, Baltimore, MD 21212, buschvi@gmail.com)

The noise in hospitals has been growing monotonically since at least 1960, and noise is now a top complaint of patients, staff, and visitors. Hospital noise sources are of many types, including HVAC noise from required high air flows, equipment noise from machines such as MRI units, alarms from equipment at patient bedsides, pneumatic tube lab transport systems, PA systems, and speech absolutely everywhere. In 2006, the federal government introduced the Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) as a standardized survey to measure patient perception of the quality of care received. Results of the survey are publicly available for each of the over 5500 hospitals in the United States. The first analysis of results showed that the lowest score received by U.S. hospitals in aggregate was the single acoustics question which asks whether patients found their room sufficiently quiet to allow for sleep at night. With the threat of decrease of federal compensation unless they show improvement, hospitals have been developing and implementing noise control programs. In this talk, we will review what we know about hospital noise, what we still need to investigate, and the challenges to achieving transformative change.

Session 3pIDb

Interdisciplinary: ASA Hunt Postdoctoral Research Fellows: Through the Years (Poster Session)

Logan Hargrove, Cochair
Marshall, VA

Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816

All posters will be on display from 1:15 p.m. to 3:15 p.m.

Invited Papers

3pIDb1. Looking back forty years on the significance of my Hunt fellowship. Steven L. Garrett (Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

A postdoctoral fellowship comes at a very important point at the start of a young scientist's career. Since the Hunt Fellowship provides salary and covers incidental expenses, it allows the applicant to select where (s)he will work and with whom. Although it was impossible to know at the time, my year as a Hunt Fellow at the University of Sussex, doing ultra-low temperature acoustics in superfluids, determined the direction of a career in acoustics that has been challenging and fulfilling over four decades. In Sussex, I shared an office with Prof. Richard Packard, on sabbatical from UC-Berkeley. He invited me to spend the following two years in his laboratory where I also had the pleasure of working with Greg Swift. When Greg left to take the Oppenheimer Postdoctoral Fellowship at Los Alamos National Laboratory to work on novel heat engines with John Wheatley, the field of thermoacoustics was born. After doing a Ph.D. degree under the supervision of Izzy Rudnick and Seth Putterman, the Hunt allowed me to continue working with amazingly talented and innovative physicists. Seth and Izzy are a hard act to follow. Having the access that the Hunt provided set me on a path of lifelong learning.

3pIDb2. From acoustic microscopy to quantum computing. Daniel Rugar (IBM Res., 650 Harry Rd., San Jose, CA 95120, rugar@us.ibm.com)

Receiving the 1981 Hunt Postdoctoral Research Fellowship helped launch a fulfilling scientific journey that still continues today. The fellowship enabled me to continue my work on gigahertz frequency acoustic microscopy at Stanford University under Professor Calvin Quate. During my fellowship I succeeded in building a microscope that focused 3 GHz acoustic waves into liquid helium at temperatures below 100 mK. Amazingly, the system actually worked, and imaging resolution below 100 mm was achieved! From this experience I developed a fascination for nanoscale imaging techniques that has guided my Research career. After leaving Stanford I joined IBM Research where I have worked on scanning tunneling microscopy (STM), atomic force microscopy (AFM), magnetic force microscopy (MFM) and, most recently, two different approaches to nanoscale magnetic resonance imaging (nanoMRI). My scientific evolution continues: I now manage a research group focused on superconducting quantum computing devices. Although my scientific journey has taken me far from the world of acoustics, I credit the Hunt Fellowship for giving me a key early opportunity for scientific exploration.

3pIDb3. 1983–1984 Hunt Postdoctoral Research Fellowship in Bergen, Norway. Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

The 1983–1984 Hunt Postdoctoral Research Fellowship in Acoustics took me to the Mathematics Institute at University of Bergen in Norway. My acoustics education began at Columbia University with an undergraduate acoustics course from Cyril Harris, who encouraged me to pursue a Ph.D. degree in Acoustics at Pennsylvania State University. My doctoral advisor at Penn State, Francis Fenlon, introduced me to nonlinear acoustics. Frank succumbed to cancer at age 41, and David Blackstock invited me to Applied Research Laboratories at University of Texas at Austin to complete my Ph.D. degree in his nonlinear acoustics group while remaining a Penn State student. At Texas I encountered another two leaders in nonlinear acoustics, Jacqueline and Sigve Tjøtta on leave from University of Bergen, who invited me to spend my Hunt Fellowship year with them in Norway. I thus had the good fortune to be instructed and mentored by pioneers who approached nonlinear acoustics from three different perspectives: engineering (Fenlon), physics (Blackstock), and mathematics (the Tjøttas). In this way, the Hunt Fellowship contributed substantively to rounding out my postgraduate education both academically and culturally. Following a second postdoctoral year at Texas I accepted a faculty appointment in Mechanical Engineering that continues to this day.

3pIDb4. F.V. Hunt and my travels with fricative consonants. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

My training as an electrical engineer aligned well with the way acoustics of speech was modeled, as I learned when working at Bell Laboratories and then in Ken Stevens' lab at MIT. However, fricative consonants require turbulence noise for their production; studying them for my Ph.D. led me toward aeroacoustics and testing whether circuit models really were adequate for modeling fricatives. The 1984–1985 Hunt and the NATO 1985–1986 postdoctoral fellowships gave me two years in which to learn more. I studied flow measurement methods at MIT's Mechanical Engineering Department (supervised by Prof. Richard Lyon), then used them in experiments on mechanical models of fricatives and of vocal folds at the Institute of Sound and Vibration Research (ISVR) at the University of Southampton, UK (supervised by Dr. Stephen Elliott and Dr. Phil Nelson). At the Department of Speech Communication and Music Acoustics at KTH, Stockholm, Sweden, supervised by Prof. Gunnar Fant, I analyzed human speech and experimented with vocal tract imaging. As academic staff at the University of Southampton I collaborated with ISVR and KTH colleagues and others on the aeroacoustic aspects of speech, vocal tract imaging and speech analysis methods; since 2004 I have continued this research at Haskins Laboratories.

3pIDb5. A career in acoustics nucleated by the 8th Hunt Fellowship (1985–1986). Anthony A. Atchley (Graduate Program in Acoust., Penn State Univ., College of Eng., 101C Hammond Bldg., University Park, PA 16802, atchley@psu.edu)

The 8th Hunt Fellowship enabled the author to work with Professor Robert Apfel and his inspirational research group at Yale University. The proposed research was to use superheated droplets to detect cavitation generated by short pules of megahertz-frequency ultrasound. At the time, it was thought that such cavitation would not produce large enough acoustic signals to be detected directly. Instead, if cavitation occurred on or near superheated droplets, they might erupt and emit a detectable signature. However, as happens in research, this idea was eclipsed by the discovery of a simpler approach. Working closely with Leon Frizzell, on sabbatical leave from the University of Illinois at Urbana-Champaign, Christy Holland, Sameer Madanshetty, and Ron Roy, we noticed that cavitation could be detected passively and used the method to investigate the dependence of cavitation thresholds on various pulse parameters. Also while supported by the Fellowship, the author was introduced to thermoacoustics which consumed the next phase of his career. This interest in high amplitude sound evolved over time into studies of the nonlinear propagation of noise from sub- and supersonic aircraft. [The author gratefully acknowledges support from the Office of Naval Research for much of the research discussed in this poster.]

3pIDb6. Music perception and the Hunt Fellowship. Ian M. Lindevald (Physics, Truman State Univ., 100 East Normal St., Kirksville, MO 63501, lindy@truman.edu)

I was the 1987–1988 Hunt Postdoctoral Research Fellow a lifetime ago. As a mathematics/physics nerd with a love of music, I consider it the greatest privilege of my life to have spent my formative years at the feet of Dr. Arthur Benade at Case Western Reserve University, where I earned my Ph.D. in physics. Art was my mentor and my friend, and I was his last Ph.D. student. I accompanied him to many ASA meetings in the early 1980's. Although Art's primary expertise was in the physics of wind instruments, he put me to work on issues of perception of musical sounds in the random sound fields of enclosed spaces. The focus on music perception paved the way for my post-doctoral research in Munich, supported by the Hunt Fellowship, where I worked with Professor Ernst Terhardt at the Technical University of Munich, testing his robust pitch algorithm with genuine musical sounds derived in a reverberant space. The results of that research mainly supported Terhardt's work. Since then, I have redirected my interests to physics education and undergraduate research and also serve in departmental leadership at Truman State University, a public liberal arts institution in rural northeast Missouri.

3pIDb7. Hunt Magic: A lifetime of fruitful collaboration. E. Carr Everbach (Eng., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu)

The 1989–1990 F.V. Hunt Fellowship allowed me to spend one year at the University of Rochester in Edwin Carstensen's lab to investigate the physical mechanisms of shock wave lithotripsy (ref Boston). Coming from the lab of Bob Apfel at Yale, where I had worked on nonlinear propagation and measurement of B/A, at U. Rochester I applied these concepts to shock wave propagation and measurement in lithotripsy. I also investigated the role of inertial cavitation in both lithotripsy and blood clot dissolution, developing collaborations with Mort Miller's biophysics lab and clinicians at the Strong Memorial Hospital. The fruitful collaborations I developed at U. Rochester during my Hunt year continued for years thereafter, including several summers spent in Rochester until marriage and family intervened. I can credit the awarding of my NSF Presidential Faculty Fellowship and decades of ultrasound bioeffects work to my Hunt Fellowship year, and I relish ongoing collaborations that enrich my teaching and research at an all-undergraduate institution.

3pIDb8. From Berlin to Atlanta, with a sojurn in Oz. Kenneth Cunefare (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Having been lured away from Exxon to pursue a Ph.D. by Gary Koopmann, I had no idea during a 1989 visit to the TU in Berlin, with the Berlin Wall still a looming presence, that he was laying the ground work with Herr Dr. Prof. Manfred Heckl of the Institute of Technical Acoustics to host me should I receive the Hunt. When I arrived in July 1990, the wall had fallen, and great changes were underway. The year passed all too quickly, with Pink Floyd at "The Wall," technical visits to Dresden, Gottingen, and ISVR, and more. The work I did in that year remains some of my highest cited work. Having already accepted a faculty position at Georgia Tech made the year much less stressful. The connections and colleagues I made at the TU remained throughout my career. At GT I explored a number of areas triggered by my Hunt year, including structural acoustic optimization, acoustic mode representation for exterior fields, and active structural acoustic control which led to a sabbatical year, with results akin to my Hunt year, at the University of Adelaide in Australia. I am also a founding member of a successful acoustic consultancy (Arpeggio).

3pIDb9. An indispensable detour: From a Hunt Postdoctoral Research Fellowship to microelectronics packaging. Quan Qi (Apple SEG Packaging, 1 Infinity Loop, MS 34-4HWT, Cupertino, CA 95014, quan_qi@apple.com)

After receiving my Ph.D. degree in Theoretical and Applied Mechanics from University of Illinois at Urbana-Champaign, I faced a choice of taking an industrial job offer or pursuing research in nonlinear acoustics as the 1992-93 Hunt Postdoctoral Research Fellow. My argument at the time was rather simple: I can always choose to go back to industry at any time but there will not be another "shot" to be a Hunt Fellow! Having studied fluid mechanics, it was a natural extension to leverage that background to focus on nonlinear acoustics for the Hunt Fellowship, specifically to investigate feasibility of using high-frequency acoustical streaming to replace the cavitation based mechanism to clean micron-sized particle from the surface of silicon wafers. During the Hunt Fellowship, I became interested in electronics packaging: wafer cleaning led to silicon chips, which in turn led to microchip packaging. I have worked in this particular area, starting at Hewlett-Packard, with other companies and now with Apple. Occasionally, I wonder: what if I had taken that industry job instead of the Hunt Fellowship? I am thankful, though, for what I have and very grateful for the Hunt Fellowship opportunity that led to gratifying professional career in microelectronics packaging.

3pIDb10. Nucleation of a career in biomedical ultrasound. T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

I was introduced to the Acoustical Society of America by Allan Pierce, who supervised my doctoral dissertation "Physical Theory of Narrow-Band Sounds Associated with Intracranial Aneurysms" in Penn State's Graduate Program in Acoustics. Professor Pierce also introduced me to Robert Waag, whose Diagnostic Ultrasound Research Laboratory I joined as a postdoctoral research associate at the University of Rochester. I received the 17th F. V. Hunt Postdoctoral Fellowship in 1994/1995 to investigate "Ultrasonic scattering: new techniques for measurement, analysis, and imaging." As a Hunt Fellow in Professor Waag's laboratory, I was immersed in the broad, interdisciplinary field of biomedical ultrasound, including tissue characterization from scattering measurements, breast imaging by diffraction tomography, and numerical modeling of propagation through realistic tissue models. My subsequent career has focused on biomedical applications of ultrasound, including positions at Penn State working with David Swanson, at Ethicon Endo-Surgery where Inder Makin recruited me to collaborate on a therapeutic ultrasound platform, and at University of Cincinnati where Christy Holland recruited me to join the Department of Biomedical Engineering. Through collaborations with these and many other acousticians, I have learned the value of applying physical acoustics principles to biomedical problems.

3pIDb11. Shock waves in the atmosphere, the body, and mercury. Robin Cleveland (Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, robin.cleveland@magd.ox.ac.uk)

As the 1995/96 F.V. Hunt Fellow, I transferred skills acquired during my Ph.D., at the University of Texas at Austin, in developing a numerical models of sonic boom propagation in the atmosphere, in order to investigate how shock waves propagate through the body, in order to better understand lithotripsy the medical procedure where shock waves are used to fragment kidney stones, at the Applied Physics Laboratory at the University of Washington in Seattle. After my fellowship, I joined the faculty at Boston University where a collaboration with Oak Ridge National Laboratory resulted in a project to understand shock wave generation and propagation in liquid mercury. This talk will focus on the underlying nonlinear acoustics associated with shock waves in fluids and by use of nondimensionalization demonstrate the strong similarity in these three apparently disparate propagation problems. It will be then shown that even with modern computers the memory and calculation requirements associated with fully three-dimensional high-fidelity models of these problems are substantial. Finally, the differences in the problems due to boundary conditions and medium properties will be discussed. The Hunt Fellowship was the vehicle by which my training in shock propagation could be used to further research in other areas.

3pIDb12. Distinctive Features. Mark A. Hasegawa-Johnson (Elec. and Comput. Eng., Univ. of Illinois, 405 N Mathews, Urbana, IL 61801, jhasegaw@illinois.edu)

I was the Hunt Post-Doctoral Fellow in 1996–1997; my Hunt fellowship led directly to a successful application for an NIH R01 post-doctoral fellowship with the same mentor, which led, in turn, to my current position as Professor at the University of Illinois. As part of my Hunt post-doc, I rented a very low frequency electromagnetic field meter, and demonstrated that the articulograph presents minimal risk to human subjects; this was published as my first article in JASA. As part of my NIH post-doc, I acquired MRI of static vowel configurations (http://isle.illinois.edu/sst/data/mri/), and demonstrated that a PARAFAC rank-3 factor analysis of the English vowels supports two speaker-independent factors matching the distinctive features. As part of my research in Urbana, I have developed automatic speech recognizers using distinctive feature models grounded in both speech production and speech perception. In 2015, Preethi Jyothi and I published the idea of mismatched crowdsourcing: an American listening to Uyghur is modeled as a machine who transcribes each distinctive feature shared by the two languages, with an error probability related to the similarity with which the two languages use the feature.

3pIDb13. An unexpected but fulfilling path: From a Hunt Postdoctoral Research Fellowship in Denmark to academia in Nebraska. Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

As the 1998-99 Hunt Postdoctoral Research Fellowship recipient, I conducted postdoctoral research at the Technical University of Denmark under the supervision of Prof. Anders Christian Gade. My goal as a teenager was to become an acoustical engineer who designed concert halls, after reading about such a job from a high school physics textbook. After receiving my Bachelor's of Science degree in Civil Engineering, I had planned to get a Master's degree in Acoustics from the Pennsylvania State University and then become an acoustical consultant; I certainly had no intention of pursuing a Ph.D., a postdoc, or an academic position. I was fortunate, though, to have had mentors who guided me on an unexpected but fulfilling career path: first in getting my Ph.D. instead of Master's degree, then applying and receiving the Hunt Postdoctoral Research Fellowship to study abroad in Europe during which I made firm connections to other strong architectural acoustics groups, and finally becoming a faculty member at the University of Nebraska's Architectural Engineering program, focused on architectural acoustics teaching and research. This poster explores the career path I have followed and reflects on the influential role the Hunt Postdoctoral Research Fellowship had upon it.

3pIDb14. An appreciation of the benefits of a Hunt Fellowship. James C. Lacefield (Elec. & Comput. Eng., Western Univ., Thompson Eng. Bldg, Rm. 279, London, ON N6A 5B9, Canada, jlacefie@uwo.ca)

I was awarded the Hunt Fellowship in 2000-01 as a postdoctoral researcher at the University of Rochester working under the supervision of Robert Waag. My research during the Fellowship experimentally analyzed methods of compensating for focus aberration in medical ultrasound imaging using multirow and matrix transducer arrays. I was subsequently recruited to a faculty position at Western University in London, Ontario, at the time Western established its Biomedical Engineering program, which I currently direct while continuing to conduct research in biomedical ultrasound. One of the enduring benefits I acquired during my Fellowship was to develop my skill at and appreciation of precise scientific communication, which are characteristics Professor Waag instilled in all of his trainees. Another valuable benefit the Fellowship granted me was an opportunity to begin forming my own scientific network through the ASA and other organizations. Holding the Hunt Fellowship provided me both encouragement and financial resources to become more active in the biomedical ultrasonics research community, which was a key step toward launching my academic career.

3pIDb15. Expect the unexpected: A path enlightened by the Hunt Fellowship. Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover W225, Athens, OH 45701, leec1@ohio.edu)

My interest in language led to an interdisciplinary and international journey through psychology (B.S., National Chengchi University, Taiwan), linguistics (A.M., Brown), cognitive science (Ph.D., Brown), and speech communication (postdoc, MIT) before joining the faculty of Communication Sciences and Disorders at Ohio University. Being awarded the Fellowship in 2001 was a pleasant surprise. The generous support of the Fellowship allowed me to witness how great scientists think, act, and educate. Studying with Professor Ken Stevens and members of the MIT Speech Group further prepared me for research on crosslinguistic aspects of speech communication. The experience also inspired me to mentor my students as I was mentored: with respect and genuine interest. As Professor Hunt envisioned, this Fellowship allowed me to play a small but meaningful part in furthering the science of, and education in acoustics. I am deeply grateful to Professors Sheila Blumstein (Brown), Phil Lieberman (Brown), and Ken Stevens (MIT) for making the journey possible and for believing in me.

3pIDb16. Hunt-ing for new therapies: From red blood cells to acoustic nanobubbles. 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)

The Acoustical Society has undoubtedly nurtured me through all key stages of my career. I attended my first ASA meeting as the last doctoral student of noise acoustician Prof. Shôn Ffowcs Williams, working on acoustic scattering from red blood cells to detect hemolysis. At a time when I was questioning whether research in general and acoustics in particular were right for me, the unexpected award of the biomedical acoustics student prize (Atlanta, 2000) and a chance meeting with my future Hunt fellowship mentor, Prof. Ron Roy, radically changed my outlook. Until that point, my perception of biomedical acoustics had been confined to diagnostic applications, but discovering the emerging therapeutic potential of focused ultrasound and acoustic cavitation made me realize that it was possible for an acoustician to impact key aspects of human health, such as treating cancer or facilitating orthopedic surgery. Much of the training and inspiration of my Hunt fellowship year at Boston University (2002–2003) travelled back with me to Oxford when I set up the BUBBL research group, which over a decade and following further encouragement by an ASA R. Bruce Lindsay award (2012) enabled my team and me to translate novel acoustic therapeutic technologies into clinical practice.

3pIDb17. HUNTing for new biomedical ultrasound applications. Tyrone M. Porter (Boston Univ., 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

The use of ultrasound in biomedical applications has evolved dramatically in the last few decades. Biomedical ultrasound first gained widespread use clinically as an imaging modality for monitoring pregnancy and diagnosing cardiovascular diseases. The introduction of encapsulated microbubbles as contrast agents enabled the development of contrast-enhanced diagnostic ultrasound initially and ultrasound-mediated drug delivery more recently. During my year as a Hunt Postdoctoral Fellow, I worked on the development of submicron vesicles that could serve as both ultrasound contrast agents and drug carriers. These vesicles could be leveraged for image-guided ultrasound-mediated drug delivery for various medical conditions. My tenure as a Hunt Postdoctoral Fellow paved the way for a faculty position at Boston University where I continue to work on vesicles and droplets in an emerging research field that combines therapy and diagnostics more commonly known as theranostics. This poster illustrates research and activities that were spawned by my tenure as a Hunt Postdoctoral Fellow.

3pIDb18. The Hunt Postdoctoral Fellowship: Opening doors to interdisciplinary research. Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Architectural Eng., Omaha, NE 68182-0816, eryherd@unl.edu)

I was honored to receive the 2006–2007 F.V. Hunt Postdoctoral Fellowship to conduct research at the Sahlgrenska Academy of Medicine at Gothenburg University in Sweden. I worked under the supervision of Dr. Kerstin Persson Waye, studying the impact of noise in hospitals on staff and patients. My background and training leading up to my postdoc was in engineering with a specialty in architectural acoustics and noise control. Through the Hunt Fellowship, I was able to deepen my skillset by pursuing interdisciplinary research in occupational and environmental health. I will never forget the excitement of working in an entirely different environment and learning how to navigate both medical terminology and another language simultaneously. The adaptability I gained by facing those challenges served me well on my path to tenure. One of the most fruitful parts of my postdoc from a research standpoint was identifying the many areas of hospital acoustics that needed more work. To this day, I continue to collaborate with students and colleagues in engineering, medicine, and other diverse disciplines to improve healthcare built environments. Ten years later, I can truly say that the F.V. Hunt Fellowship had a profound impact on my life and career path.

3pIDb19. The Hunt Fellowship: Inspiring acoustics professionals of the future. Alison K. Stimpert (Bioacoustics/Vertebrate Ecology, Moss Landing Marine Labs., 8272 Moss Landing Rd., Moss Landing, CA, alison.stimpert@gmail.com)

Students may exit a Ph.D. program feeling fatigued, and sometimes a bit directionless professionally. These feelings combined with the challenging academic job market could be part of the reason for this time period being one of the biggest "leaks in the pipeline," for women scientists in particular. The flexibility and generosity of the Hunt Fellowship can be inspiring at this crossroads of a young scientist's career. As the 2009–2010 Hunt Postdoctoral Research Fellowship recipient, I studied the acoustic behavior of humpback whales on their Antarctic feeding grounds, working under the mentorship of Dr. Whitlow Au (University of Hawaii) and Dr. Douglas Nowacek (Duke University). This project followed seamlessly from my dissertation, yet brought me to new ecosystems and reinvigorated my research questions with surprising data. The signal processing skills I gained in acoustic and animal movement analysis solidified my interest in quantitative acoustic behavior research and I made strong connections with several interdisciplinary research groups. This all fueled my career path toward my current Research Faculty position at Moss Landing Marine Laboratories, using similar sound and movement tags to study the effects of anthropogenic noise on cetaceans in general.

3pIDb20. The French connection: My experience as a Hunt Fellow. Michael Canney (CarThera, 1490 Delgany St., Apt. 603, Denver, CO 80202, michael.canney@gmail.com)

After finishing my Ph.D. degree in 2009 at the University of Washington, I was grateful to be the recipient of the Hunt Fellowship. During my year as a Hunt Fellow, I worked at an academic laboratory in Lyon, France, under the supervision of Dr. Cyril Lafon and Dr. Jean-Yves Chapelon. The laboratory specialized in the field of therapeutic ultrasound and had several translational projects to take new therapeutic ultrasound technologies into the clinic. Little did I realize that a one-year fellowship would turn into a 6-year hiatus living abroad in France and a working relationship with the French that continues to this day. After my Hunt fellowship, I continued as one of the first employees of a French startup company and helped lead the development of a novel therapeutic ultrasound device for brain drug delivery. During the six years that I worked in France, this device was successfully transitioned from animal experiments to clinical trials. This poster explores the work that I started in France on these technologies and which I continue to work on presently. In addition to the professional development it allowed, the Hunt Fellowship gave me the opportunity to pursue a lifelong dream to live and work in a foreign country and to become fluent in a second language.

3pIDb21. In pursuit of noninvasive microsurgery. James J. Choi (Dept. of BioEng., Imperial College London, Rm. 4.06, Royal School of Mines Bldg., South Kensington Campus, London SW7 2AZ, United Kingdom, j.choi@imperial.ac.uk)

I am committed to creating safe and high performance noninvasive surgical devices and the 2011 Hunt Fellowship supported me at a critical point on my path toward this goal. Noninvasive surgery refers to the local modulation of tissue deep in the body without surgical incisions. This technology has the potential to treat devastating diseases, such as Alzheimer's disease cancers, and stroke. I entered research in this field as an undergraduate at the University of Michigan building multi-element therapeutic arrays under Prof. Charles Cain. I then developed technologies that could deliver drugs to the brain at Columbia University during my doctoral research under Prof. Elisa Konofagou. At this stage, I noticed a severe gap in capability. Although we could spatially control where an ultrasound beam is placed, we had poor control of the safety and performance within it. The 2011 Hunt Fellowship supported me on the development of passive acoustic mapping technologies—a unique technology that allowed for the resolution of sub-beam interactions. This work was conducted at the University of Oxford under Prof. Constantin Coussios. I now lead a team at Imperial College London as a tenured Associate Professor and remain committed to continuing my pursuit of noninvasive microsurgical devices.

3pIDb22. The unexpected but fulfilling impact of the Hunt Postdoctoral Research Fellowship in my academia career. Likun Zhang (Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu)

Soon after I graduated with my Ph.D. degree in Physics from Washington State University in 2012 under the supervision of Dr. Philip L. Marston, I applied to the 2013–2014 Hunt Postdoctoral Research Fellowship. I was honored to receive the fellowship that funded my proposed postdoctoral research on acoustic propagation through internal waves under the joint supervision of Dr. Harry L. Swinney at the University of Texas at Austin and Dr. James F. Lynch at Woods Hole Oceanographic Institution. I am very grateful to the Hunt Fellowship for providing me with the opportunity to shift from analytical work on physical acoustics and fluid dynamics in my Ph.D. study to numerical and laboratory work on underwater acoustics in postdoctoral research. This has broadened the range of tools that I have to address acoustic problems in my academia career. I am fortunate to have had mentors who guided me on an unexpected but fulfilling career path that led me to becoming a faculty member in Physics at the University of Mississippi with my research affiliated with the National Center for Physical Acoustics. This poster explores the influential role of the Hunt Fellowship in the career path that I have taken.

3pIDb23. I wonder how animals can do it so well: An ongoing detour to build better sonar, enabled by the Hunt fellowship. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu)

The Hunt Postdoctoral Research Fellowship gave me the freedom and flexibility to take a detour to pursue research topics that are not on a linear path from my Ph.D. training. My graduate study in the MIT-WHOI Joint Program in Oceanography focused on modeling and measuring the acoustic scattering of marine organisms for improving sonar, or echosounder, performance. Knowing the current limitation of human-made sonar, I have always been curious about how echolocating bats and dolphins can perform acoustic-guided search, identification and tracking of prey so efficiently. As a Hunt fellow in Dr. Cynthia Moss' lab in Johns Hopkins University, I learned first-hand the sophisticated echolocation and flight behavior of bats, which helped me develop insights for computational modeling. Using the research funds provided by the Hunt fellowship, I was fortunate to work with Dr. Whitlow Au, who first introduced me to the field of acoustics before graduate school, again to investigate the dolphins' adaptive biosonar beam control during prey capture. These opportunities lay the foundation for me as an early career scientist to integrate the conventional physics-based approach with the study of biological sonar toward a goal of developing better acoustic tools to observe and understand the ocean.

3pIDb24. Physical Acoustics and Oxford: My experience as a researcher, a fellow, and beyond. Jason L. Raymond (Dept. of Eng. Sci., Univ. of Oxford, 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, jason.raymond@eng.ox.ac.uk)

In May 2015, I arrived in London after completing my Ph.D. work under the direction of Prof. Christy Holland at the University of Cincinnati and moved shortly thereafter with a goal to help establish one of the principal physical acoustics laboratories in the UK at Oxford. At the outset, it was my hope and expectation that the F. V. Hunt Fellowship would help to define a unique trajectory for my career. During the fellowship and along with the support of my mentor, Prof. Ronald Roy, I have been able to develop a network of collaborators and conduct research which will help to define the next phase of my career and work in acoustics. A highlight of my term as a Hunt Fellow was the opportunity to travel to China, during which I visited four research laboratories relating to my work in therapeutic ultrasound and delivered two invited seminars. I am grateful to the Hunt family estate and the Acoustical Society of America for offering this fellowship, which has not only provided me with the opportunity to conduct research, but also the flexibility to develop as an independent scientist and to work with international collaborators.

3pIDb25. Impact of the F. V. Hunt postdoctoral fellowship on a trainee's research and career advancement. Himanshu Shekhar and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu)

The F. V. Hunt Postdoctoral Research Fellowship in Acoustics gave Dr. Himanshu Shekhar an opportunity to pursue a new research direction in therapeutic ultrasound at University of Cincinnati with Prof. Christy Holland as a mentor. This research was performed at the Image-Guided Ultrasound Therapeutics Laboratories, which is composed of trainees, principal investigators, physician-scientists, and clinical collaborators. During his tenure as a Hunt fellow (June 2015–May 2016), Dr. Shekhar developed cavitation-mediated lytic and bioactive gas delivery to treat ischemic stroke. He also contributed to ongoing research on microfluidic manufacture of thrombolytic-loaded echogenic liposomes, thrombolysis using histotripsy, and comparative lytic activity of a lytic in human and porcine clots. His work led to two first author peer-reviewed manuscripts, three coauthored manuscripts, and two presentations at the meetings of the Acoustical Society of America (ASA). This fellowship also enabled Dr. Shekhar to enhance his engagement with the ASA as a full member of ASA, serving on the biomedical acoustics technical committee, and on subcommittees of the live-streaming initiative and the Hunt recognition campaign. The training, mentorship, and the exposure received during the Hunt Fellowship tenure has prepared Dr. Shekhar, who plans to pursue an academic research career in biomedical acoustics.

3pIDb26. The F.V. Hunt fellowship: A step toward research independence and leadership. Romain Fleury (EPFL, EPFL - STI - LWE, ELB 033 - Station 11, Lausanne 1015, Switzerland, romain.fleury@epfl.ch)

As the recipient of the 2016 F.V. Hunt postdoctoral fellowship, I had the immense privilege to experience postdoctoral research in one of the most active acoustic groups in France, in the Langevin Institute at ESPCI (School of Physics and Chemistry) under the supervision of Prof. Mathias Fink and Dr. Geoffroy Lerosey. This poster presents what I learned during this wonderful experience and the impact of the fellowship on my academic career and identity as a researcher.

3pIDb27. Frederick V. Hunt Postdoctoral Research Fellowship: My journey toward clinical research in audiology. Anna Diedesch (Dept. of Otolaryngology/Head & Neck Surgery, Oregon Health & Sci. Univ., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, acdiedesch@gmail.com)

As the current Hunt Postdoctoral Research Fellow, I am evaluating binaural cue distortion in hearing aids at the VA RR&D National Center for Rehabilitative Auditory Research. Beginning early in life I always wanted to teach. Ironically, during my undergraduate education I felt that would require too much education. I instead found the field of Audiology and was fascinated by the science and technology in the field as well as rewarded by being able to treat and counsel patients regarding their hearing difficulties. While working on my Doctor of Audiology, I became aware of the difficulties in our field to train clinical researchers who would return to academia to train the next generation of audiologists. I was hesitant to pursue a Ph.D., so instead took a job as a Research Audiologist while I figured out my next step. With the guidance from Frederick J. Gallun and Marjorie Leek, I eventually began my Ph.D. with G. Christopher Stecker. This poster explores my journey toward becoming a clinical researcher and how the Hunt Fellowship is influencing my career trajectory.

Session 3pNS

Noise and ASA Committee on Standards: Urban Planning Using Soundscape I

David Woolworth, Cochair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

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

Contributed Papers

1:15

3pNS1. Acoustic planning of a music festival site. Adam Young (CSTI Acoust., 16155 Park Row, Ste. 150, Houston, TX 77084, adam@cstiacoustics.com)

CSTI acoustics worked with a community on the planning of a festival site that will host outdoor concerts. Our work involved measuring sound from an outdoor concert at a nearby venue, documenting ambient sound levels in communities surrounding the proposed festival site, modeling the sound propagation from optional stage locations and orientations, and proposing procedures and sound limits for future performances.

1:30

3pNS2. A big dilemma: Airport noise and property value. Yalcin Yildirim (Urban Planning and Public Policy, Univ. of Texas at Arlington, 601 W Nedderman Dr. #203, Apt. 933, Arlington, TX 76019, yalcin.yildirim@mavs.uta.edu) and Sriram Villupuram (Finance and Real Estate, Univ. of Texas at Arlington, Arlington, TX)

Even though many attributes to the airport creates value for properties, many other factors such as noise is considered to have wither neutral or negative effects residential property values. This research examines the relationship between airport noise and housing prices by using disaggregated transactions in Dallas-Fort Worth Airport which is the fourth busiest airport in the United States (FAA, 2015). The effects of airports on residential property values have been studied in several fields and contexts. Tomkins *et al.* (1998) examined the Manchester Airport by performing hedonic method. In addition, Nelson conducted a meta-analysis about noise and housing price near airport and these studies applied census tract aggregated data. This paper examined the implications of the airport noise on disaggregated residential property transactions in a rapid growing urban area—Dallas Fort Worth Metropolitan- by performing Multi-Level-Modeling. To do this, variables are defined for each level. Property level is considered with many characteristics such as number of bedrooms, bathrooms, floor, etc. Moreover, there are also neighborhood attributes such as demographics, income, and green space to evaluate. Based on results, total and marginal benefits of noise reduction were estimated that a 1 dB noise reduction may lead to increase 0.6% of house values.

1:45

3pNS3. Parking lot noise evaluation. Christopher L. Barnobi (Noise and Air Quality, Dudek, 1102 R St., Sacramento, CA 95816, cbarnobi@dudek.com), Connor Burke (Noise and Air Quality, Dudek, Encinitas, CA), and Jonathan Leech (Noise and Air Quality, Dudek, Santa Barbara, CA)

Parking Lot noise is regularly evaluated for CEQA noise assessments, but a standardized method has not been generally adopted. This paper presents sound level measurements of common parking lot activities. Other published sound level data for parking lot activities is provided to further solidify typical parking lot sound levels. A review of published methods for evaluating parking lot noise is summarized. The appropriateness of the methodology is evaluated and a simplified methodology is presented based on the similarities and differences applicable to California for CEQA analysis and the United States in general.

2:00

3pNS4. Electric vehicles and environmental noise: Assessing the noise impact of an electric fleet through strategic noise mapping. Eoin A. King (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

Electric vehicles are fast becoming a reality and are being heralded as a real alternative to the highly polluting internal combustion engine fleet. Often electric vehicles are reported as being silent vehicles and may significantly reduce a population's exposure to environmental noise. This paper investigates what effect the widespread adoption of an electric fleet would have in a midsize city in the United States. The source level of an electric fleet is estimated by a combination of previous pass-by measurements, published data, and calculation. These estimations are used to develop a noise map of the city assuming an electric fleet. Results are then compared to noise levels assuming the current fleet. Results show that while an electric fleet will improve environmental noise levels within a city, the overall benefit is limited when expressed using a traditional strategic noise map, primarily due to the use of a time averaged L_{eq} indicator. In light of this, alternative metrics to better account for the changing acoustic characteristics of electric vehicles are discussed.

2:15

3pNS5. A preliminary look at the acquisition of acoustic field data using selected smartphone apps. Jennifer Lentz (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu) and Roger M. Logan (Houston, TX)

Numerous smartphone applications exist that measure sound levels in the acoustic field. In some cases, these apps also provide additional acoustic data in the form of real-time spectral analysis. Yet, very few studies exist that have evaluated the validity of these apps for sound level measurement. Notably, these studies evaluated performance in the laboratory and only assessed the accuracy of the measurement of sound levels. Here, we present field-acquired data on the accuracy of dB level measurement and spectral analysis of a variety of smartphone apps for both Android and iOS platforms. We then discuss the feasibility of using these applications for field data acquisition.

Session 3pPA

Physical Acoustics: General Topics in Physical Acoustics II

Alexey S. Titovich, Chair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

Contributed Papers

1:00

3pPA1. Acoustic and seismic propagation between water and land—An experimental study. Michelle E. Swearingen (Construction Eng. Res. Lab., CERL, US Army ERDC, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Donald G. Albert, and Jason R. Dorvee (CRREL, US Army ERDC, Hanover, NH)

Pressure waves can travel through coupled media, such as water and land. A field experiment was performed to characterize impulsive signal propagation between these media. Balloon pops underwater provided the in-water source, while sledgehammer blows provided the land-based source. Data were collected using a distributed sensor array that included locations on land and in the water, using a combination of microphones, hydrophones, and geophones. The array extended 75 m from the shore onto land and 75 m into the water perpendicular to the shoreline. Additional sensors were placed 25 m from the nominal shoreline and parallel to the shore. Source locations varied over the area to maximize capture of varying propagation paths. Data are analyzed for travel time, amplitude, and waveform shape. A description of the field experiment and results are presented. [Work funded by U.S. Army ERDC.]

1:15

3pPA2. Seismic wave propagation between ice-covered water and land—An experimental study. Jason R. Dorvee, Donald G. Albert (CRREL, US Army ERDC, US Army Corps of Engineers, ERDC, 72 Lyme Rd., Hanover, NH 03755, Jason.R.Dorvee@usace.army.mil), and Michelle E. Swearingen (CERL, US Army ERDC, Champaign, IL)

A field experiment was performed to characterize impulsive signal propagation between ice-covered water and land in both directions. Sledgehammer blows provided the source on both the surface of the ice and the ground. Underwater signals were generated with a balloon breaker source positioned 1 m below the surface of the ice. Sources on the ice and on land were located either in-line with the sensor array or oblique to array. Data were recorded using a combination of hydrophones in the water, geophones on the ice, geophones on the land, and hydrophones on land used as microphones. The different velocity structure in the two media produces different waveform characteristics. A hammer blow on ice produces a wave about three times larger on ice compared to one on the ground. In general, waves traveling in the ice have higher frequency content, lower attenuation, and travel faster than waves on land. In addition, the water layer below the ice produces reverse dispersion with higher frequencies arriving before lower frequencies, the opposite of the situation on land. As these waves transit the shoreline, complex waveform changes occur as a result of the different propagation characteristics. [Work funded by U.S. Army ERDC.]

1:30

3pPA3. Modeling particle-laden and poroacoustic flow phenomena via the generalized continua approach. Pedro M. Jordan (Acoust. Div., U.S. Naval Res. Lab., Stennis Space Ctr., MS 39529, pedro.jordan@nrlssc.navy.mil)

With a focus on acoustic phenomena, we investigate several dual-phase (i.e., fluid-solid) flows using the generalized continua (GC) modeling approach, where by "GC" we mean modern generalizations of the constitutive relations of classical continuum mechanics that seek to capture the impact of sub-scale structure/dynamics on the (macroscopic) field variables. Working under the finite-amplitude framework, we derive and analyze generalizations of the weakly nonlinear versions of the Euler and Navier–Stokes equations for the case of particle-laden and poroacoustic flows. Using both analytical and numerical methods, we examine the impact of the solid phase on the propagation and evolution of (1D) traveling and acceleration waves. Along the way, the advantages and disadvantages of this modeling approach will be discussed and applications to other fields noted. [Work supported by ONR funding.]

1:45

3pPA4. Introducing C/S: A companion nonlinearity indicator to the Morfey-Howell Q/S. Won-Suk Ohm, Taeyoung Park (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ohm@yonsei.ac.kr), Kent L. Gee, and Brent O. Reichman (Brigham Young Univ., Provo, UT)

The Morfey-Howell Q/S is a frequency-domain, single-point nonlinearity indicator, initially proposed for propagation of intense broadband noise [AIAA J. **19**, 986–992 (1981)]. It represents the extra change in level of a frequency component due to the nonlinearly generated absorption [*J. Acoust. Soc. Am.* **139**, 2505–2513 (2016)]. At the center of the definition of Q/S is the quadspectrum, which is the imaginary part of the cross-spectrum between the pressure and squared-pressure waveforms. The real part of the cross-spectrum (known as the cospectrum), however, has received little attention until recently Ohm *et al.* showed that its normalized version, hereby dubbed C/S, reflects the extra change in phase angle of the frequency component due to the nonlinearly generated dispersion [Proc. Mtgs. Acoust. **29**, 045003 (2016)]. In other words, Q/S and C/S signify two sides of the same coin in a nonlinear wave process under an arbitrary absorption/dispersion law: C/S is to dispersion as Q/S is to absorption. In this talk, we demonstrate the use of C/S in comparison with Q/S for the case of finite-amplitude waves in a thermoviscous fluid with multiple relaxation mechanisms.

2:00

3pPA5. Dynamics of quasi-empty rupture in the layer of cavitating liquid under SW-loading. Valeriy Kedrinskiy (Physical Hydrodynamics, Lavrentyev Inst. of Hydrodynamics, Russian Acad. of Sci., Lavrentyev Prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru) and Ekaterina Bolshakova (Physical Dept., Novosibirsk State Univ., Novosibirsk, Russian Federation)

The problem of formation and collapse of a quasi-empty rupture in the layer of a cavitating liquid under shock wave (SW) loading is considered. The SW-pulse is generated in the layer by electro-magnetic hydrodynamic shock tube in a result of high-voltage discharge of capacitor bank on a flat helical coil. The latter is located right up to the bottom (conducting membrane) of container with the layer of two-phase distilled liquid. The analysis of the experimental data shows that the rupture is shaped as a spherical segment, which retains its topology during the entire process of its evolution and collapse. It was shown that potential energy of maximum volume of rupture at its collapse is practically transformed in acoustical losses (SW-radiation) and rupture disappears. Dynamics of main parameters and an existence time of rupture were determined. The analysis of cavitating nuclei state in the form of thin layer on an entire interface of rupture shows that in the field of rupture collapse the thin cavitating layer is transformed to a cavitating cluster. The latter takes the form of a ring-shaped bubbly vortex floating upward to the free surface of the liquid layer. A p-κ two-phase mathematical model was formulated, and calculations were performed to investigate the collapse of a quasi-empty spherical cavity (rupture model) in the unbounded cavitating liquid, generation of ultra-short shock wave and to discover the dynamic growth of micro-bubbles in a cluster by five orders of magnitude. [Work supported by RFBR Grant 15-05-03336.]

2:15

3pPA6. Enhancing the convergence of multipole expansions at intermediate frequency. Hui Zhou, Emily Mui, and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, hui_zhou@student.uml.edu)

In this work, we examine near-field acoustic wave scattering from media having a spatial variation in compressibility contrast. Typically, the scattered acoustic pressure can be expressed as a convergent Neumann series when the compressibility contrast is relatively small. However as the magnitude of the compressibility contrast increases a resonant scattering condition ensues, yielding a divergent series. It has been shown that Padé Approximants method can be used in these cases, thereby extending the range of utility of the Neumann series solution. The proposed research explores hybrid methods that combine multipole-methods, and Padé Approximant approaches to calculate the near-field scattered pressure. As part of this work, we will examine the low-frequency breakdown in the plane wave, and monopole-monopole expansion approaches as it affects the numerical stability of spatial translation operations.

2:30

3pPA7. Application of particle-based computational acoustics to sound scattering by vortex and moving bodies. Yong Ou Zhang (Dept. of Naval Architecture, Ocean and Structural Eng., School of Transportation, Wuhan Univ. of Technol., Wuhan 430063, China, zhangyo1989@gmail.com)

The Lagrangian meshfree method with interacting particles is a powerful and natural approach for simulating physical systems with complicated domain topologies, moving boundaries, and multiphase media. Particle-based computational acoustics (PCA) is a noval branch of computational acoustics that aims to simulate acoustic phenomenon by Lagrangian meshfree particle methods. The ability of different particle methods to simulate flow-acoustic and flow-structure-acoustic interaction problems is evaluated, and problems include scattering of sound by a vortex or moving bodies. To separate the acoustic perturbation from the particle motion, Lagrangian acoustic perturbation equations (LAPE) including two sets of governing equations are used. Smoothed particle hydrodynamics (SPH), corrective smoothed particle method (CSPM), and finite difference particle method (FDPM) are selected for a comparison. Several checks on the accuracy and convergence of the Lagrangian meshfree PCA method are discussed. Numerical results are obtained for vortex scattering and sound wave scattering by moving rigid bodies. Various acoustic boundary conditions including moving boundaries and perfectly matched layers are examined.

Session 3pSC

Speech Communication: Clinical Populations (Poster Session)

Irina A. Shport, Chair
English, Louisiana State University, 260-G Allen Hall, Baton Rouge, LA 70803

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 3:15 p.m.

Contributed Papers

3pSC1. Measuring progress during practice: Motion analysis throughout visual biofeedback treatment for residual speech sound errors. Rebecca Mental (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, rlm142@case.edu), Holle Carey (Vulintus, Dallas, TX), Gregory S. Lee (Elec. Eng. and Comput. Sci., Case Western Reserve Univ., Cleveland, OH), Michael J. Hodge (Speech-Lang. Pathol., Cleveland Hearing and Speech Ctr., Cleveland, OH), and Jennell Vick (Psychol. Sci., Case Western Reserve Univ., Cleveland, OH)

The question of why some individuals make progress in speech therapy while others do not remains largely unanswered. Treatment delivery method could be a factor; individuals whose speech sounds have not improved through traditional therapy may be more responsive to alternative forms of treatment, such as visual biofeedback. The present study utilized visual biofeedback in the form of Opti-Speech, which uses real-time, three-dimensional streaming data from the Wave EMA system to create an avatar of a participant's tongue. Participants included two adult females with residual /r/ errors. One participant demonstrated marked improvement during and after treatment, while the other exhibited little perceptual change. It is possible that a more flexible motor system (i.e., one that shows more variability as a new skill is being learned) is more conducive to the acquisition of new speech sound movements than a more rigid system. Kinematic data were analyzed from each session, including duration, maximum displacement, distance traveled, and peak and average speeds. The coefficient of variation was calculated for each measure to assess variability. The length of the ballistic phase versus the corrective phase of movement was also calculated as treatment progressed. These measures were compared to perceptual outcomes.

3pSC2. Articulatory compensation strategies employed by an aglossic speaker. Asterios Toutios, Dani Byrd, Louis Goldstein, and Shrikanth S. Narayanan (Univ. of Southern California, 3740 McClintock Ave., EEB 400, Los Angeles, CA 90089, toutios@sipi.usc.edu)

We are employing real-time MRI to probe the speech production patterns of a speaker with congenital aglossia, a rare syndrome in which an individual is born without a tongue. The speaker being studied has only a small stump-like tongue rudiment in the region of the tongue root and a hypertrophied floor of the mouth (mylohyoid) and base of the tongue. Nevertheless, the speaker has acquired the ability to produce highly intelligible speech. One initial finding is that the speaker, in the absence of a tongue tip, produces plosive consonants that are perceptually similar to /t/ and /d/ by using a bilabial constriction with a significantly increased anteroposterior extent (relative to her /p/ or /b/ productions). Our aim is to provide an articulatory-acoustic account of this compensatory strategy via detailed analysis of her production strategies as evidenced in the real-time MRI data and via simulations using an articulatory synthesizer. We examine whether the increased anteroposterior extent of the bilabial constriction can by itself fully explain its coronal-like percept, or whether other factors contribute, such as the overall shaping of the back cavity or the dynamics of the "laminal" bilabial (pseudo-coronal) constriction formation and release. [Work supported by NIH.]

3pSC3. Subject-specific anatomical assessment of the human tongue in amyotrophic lateral sclerosis (ALS) by high-resolution MRI and diffusion tensor imaging. Euna Lee, Fangxu Xing, Sung Ahn, Timothy Reese, Ruopeng Wang (Radiology, MGH/Harvard, Boston, ME), Jordan Green (MGH Inst. of Health Professions, Boston, MA), Nazem Atassi (Neurology, MGH/Harvard, Boston, MA), Van Wedeen, Georges El Fakhri, and Jonghye Woo (Radiology, MGH/Harvard, 55 Fruit St., White 427, Boston, MA 02114, jwoo@mgh.harvard.edu)

Amyotrophic Lateral Sclerosis (ALS) is a progressive and irreversible neurological disorder, which affects upper and lower motor neurons in the motor cortex that control voluntary movements including speech and swallowing. High-resolution MRI (hMRI) and Diffusion Tensor Imaging (DTI) can provide non-invasive imaging of three-dimensional muscle anatomy and fiber myoarchitecture such as fiber orientation within the human tongue, respectively. In this work, we aim to assess anatomical differences of the tongue using both imaging methods by demonstrating the differences in quantities related to fiber connectivity for both normal and ALS subjects. We first manually delineate the genioglossus and superior longitudinal muscles on hMRI, which are aligned to each b0 image of DTI using deformable registration to provide regions of interest. We then compute fractional anisotropy and statistics about fibers connecting each pair of muscles. We apply our framework on five datasets including both normal and ALS subjects, revealing obvious quantitative degradation of muscle fibers in ALS patients compared to that in controls within and between muscles. Our framework has the potential to provide insight regarding the detrimental effects of ALS on speech and swallowing when combined with tongue motion data.

3pSC4. Acoustic speech analysis of patients with decompensated heart failure: A pilot study. Olivia Murton (Speech and Hearing BioSci. and Technol., Harvard Med. School, 24 Peabody Terrace, #1202, Cambridge, MA 02138, omurton@g.harvard.edu), Maureen Daher, Thomas Cunningham, Karla Verkouw, Sara Tabtabai, Johannes Steiner (Inst. for Heart, Vascular and Stroke Care, Massachusetts General Hospital, Boston, MA), Robert E. Hillman (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), G. W. Dec, Dennis Ausiello (Inst. for Heart, Vascular and Stroke Care, Massachusetts General Hospital, Boston, MA), and Daryush Mehta (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA)

Heart failure (HF) is a chronic condition characterized by impaired cardiac function, increased intracardiac filling pressures, and peripheral edema. HF can escalate into decompensation, requiring hospitalization. Patients with HF are typically monitored to prevent decompensation, but current

3p WED. PM

methods are of only limited reliability (e.g., weight monitoring) or invasive and expensive (e.g., surgically implanted devices). This study investigated the ability of acoustic speech analysis to monitor patients with HF, since HF-related edema in the vocal folds and lungs was hypothesized to affect phonation and speech respiration. Ten patients with HF (8 male/2 female, mean age 70 years) undergoing inpatient treatment for decompensation performed a daily recording protocol of sustained vowels, read text, and spontaneous speech. Mean length of stay was 7 days, and average weight loss was 8.5 kg. Acoustic features extracted included fundamental frequency, cepstral peak prominence, automatically identified creaky voice segments, and breath group durations. After treatment, patients displayed increased fundamental frequency (mean change 3.7 Hz), decreased cepstral peak prominence variation (mean change -0.41 dB), and a higher proportion of creaky voice in read passages (mean change 6.9 percentage points) and sentences (mean change 5.5 percentage points), suggesting that phonatory biomarkers may be early indicators of HF-related edema.

3pSC5. Not all /r/s and /l/s are the same: Classification of errors in children with cochlear implants versus normal hearing. Laura Conover and Ruth Bahr (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD 1017, Tampa, FL 33612, lconover1@mail.usf.edu)

Purpose: As children acquire speech sounds, they progress from clear substitutions, to "intermediate forms" (covert contrasts), to adult-like productions. However, there is evidence that this progression may be different in CI users as compared to NH children. Identification of intermediate forms is dependent upon rating scales that are sensitive to fine phonetic detail, such as visual analogue scales (VAS). This study uses both traditional VAS and a new 3-dimensional rating scale to explore differences in the /r,l/ productions of young children with and without hearing loss. **Methods**: Correct and error productions of /r,l/ were extracted from a standardized articulation test for nine congenitally deafened children who received cochlear implants (CIs) prior to age 3 and their speech age-matched controls. Stimuli were shortened to nonsense syllables to prevent real-word bias. Listeners rated these productions on both a VAS and a triangular scale, which allowed listeners to rate phone quality as a multi-dimensional function of /r/, /l/, and /w/. **Results & Conclusions:** Both rating scales were sensitive to subtle acoustic differences in speech sounds. Listener responses suggested more variability in sound production within the group of CI users. Listeners preferred the triangular scale because it provided greater response sensitivity.

3pSC6. Articulatory kinematics of bilateral stops in speakers with Parkinson's disease: Preliminary data. Sarah E. Worrell and Yunjung Kim (Louisiana State Univ., 85 Hatcher Hall, Louisiana State University, Baton Rouge, LA 70803, sworre2@lsu.edu)

The current study presents data on articulatory kinematics of bilateral stop production in speakers with Parkinson's disease (PD) for the long term purpose of developing a segment-specific articulatory profile of people with PD. As an initial step, we examined bilateral stops because of their relatively high frequency of occurrence and involvement of articulatory motions associated with surrounding vowels (Kim, Berry, and Kuo, 2017; Mines, Hanson, and Shoup, 1978). A total of 10 speakers (5 speakers with PD and 5 speakers without PD) were asked to read *The Caterpillar* passage in a conversational voice. An electromagnetic articulatography system (Wave, NDI) was used to track the motion of the tongue (tongue front and back) and lip (upper lip and lower lip) during bilabial stop productions (from the onset of stop closure interval to the onset of the following vocalic nuclei). The results will be presented regarding the (1) range (e.g., 2D distance) and (2) timing (e.g., the timing of minimum lip aperture) of articulatory movements focusing on group comparisons between the two speaker groups, people with and without PD.

3pSC7. Acoustic properties of vowel production in Mandarin-speaking patients with post-stroke spastic dysarthria. Zhiwei Mou (Jinan Univ., 613 West HuangPu Ave., Rehabilitation Department, HuaQiao Hospital, GuangZhou 510630, China, 405038856@qq.com), Jing Yang (Univ. of Central Arkansas, Conway, AR), Zhuoming Chen, Hong Wang, Yinhui Jiang, Jiamin Li, Wen Deng (Jinan Univ., Guangzhou, China), and Li Xu (Ohio Univ., Athens, OH)

This study investigated the acoustic features of vowel production in Mandarin-speaking patients with post-stroke spastic dysarthria. The subjects included 31 native Mandarin-speaking patients with post-stroke spastic dysarthria (age: 33–73 years old) and 40 normal adults in a similar age range (age: 22–68 years old). Each subject were recorded producing a list of 28 Mandarin monosyllables that composed of six monophthong vowels (i.e., /a, o, ɤ, i, u, y/) embedded in the /CV/ context. The patients' speech samples were evaluated by two native Mandarin speakers. The evaluation scores were then used to classify each patient into one of the two categories: mild or moderate-to-severe severity. Midpoint F1 and F2 of each vowel token were extracted and normalized. Results showed no significant differences between the patients and normal speakers on vowel duration. However, the vowel categories in the patients were more scattered and greatly overlapped than in the normal speakers. The magnitude of the vowel dispersion and overlap increased as a function of the severity of the disorder. The deviations of the vowel acoustic features in the patients from the normal speakers may provide guidance for clinical rehabilitation to improve the speech intelligibility of this type of patients.

3pSC8. Articulatory kinematics during stop closure in speakers with Parkinson's Disease. Austin R. Thompson, Amanda Kuylen, and Yunjung Kim (Louisiana State Univ., 56 Hatcher Hall, Field House Dr., Baton Rouge, LA 70803, atho184@lsu.edu)

Numerous studies have identified the perceptual characteristics of speakers with Parkinson's disease (PD), *imprecise consonants* (e.g., Darley, Aronson, and Brown, 1969). Acoustic studies have supported these findings with the observations such as spirantization, or the incomplete closure and fricative-like production of aperiodic noise during the closuresilent interval of stop consonants (Weismer, Yunusova, and Bunton, 2012). The current presentation explores the articulatory kinematics in speakers with PD during the closure interval duration of stop consonants with respect to distance, displacement, and timing of inter-articulators motion. In addition, the changes in articulatory movements during this brief time interval are examined as the speakers voluntarily vary the degree of speech intelligibility. Participants with PD and neurologically healthy controls were asked to read sentences containing stop consonants (e.g.,"*Buy Bobby a puppy*"). Movement data were collected using the WAVE (NDI, Canada). The results from the five articulatory measurement points (tongue front, tongue back, upper lip, lower lip, and jaw) will be presented with emphasis on segment-specific movement characteristics of individuals with Parkinson's disease, PD.

3pSC9. Tongue- and jaw-specific contributions to increased vowel acoustic contrast in response to slow, loud, and clear speech in talkers with dysarthria. Antje Mefferd (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 8310 Medical Ctr. East, Nashville, TN 37232, antje.mefferd@vanderbilt.edu)

Slow, loud, and clear speech can elicit increased acoustic vowel contrast in talkers with dysarthria. However, articulator-specific changes in response to these speech modifications and their relative contribution to vowel acoustic changes remain poorly understood despite the fact that these three speech modulations are commonly used as speech treatments to increase intelligibility in dysarthria. This preliminary study examined tongue and jaw movements in talkers with Parkinson's disease (PD) and amyotrophic lateral sclerosis (ALS) using electromagnetic articulography. Participants repeated the phrase "See a kite again" five times under four speech conditions: typical, slow, loud, and clear speech. Tongue movements were decoupled from the jaw to determine the relative contribution of the jaw and tongue to the overall tongue composite movement during the diphthong /ai/ in "kite". In the acoustic signal, the F2 minimum during /a/ and the F2 maximum during /i/ and their corresponding F1 values were extracted to calculate acoustic vowel contrast in F1-F2 vowel space. Linear regression analyses were used

to determine tongue- and jaw-specific contributions to acoustic vowel contrast changes in response to these speech modifications. Data analysis is currently underway. We hypothesized that talkers with PD and ALS demonstrate different response patterns to these speech modifications.

3pSC10. Tic word duration and speaking rate in Tourette's. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu)

Tourette's syndrome is a neurological condition characterized by the presence of an inventory of involuntary movements and vocalizations called tics that is unique to each individual. Tics occur on a background of typical, ongoing voluntary behavior, including speech. The voluntary speech of persons with Tourette's, like the speech of neurotypicals, shows expected signatures of underlying prosodic structure, e.g., emphatic word lengthening, utterance-final word lengthening and changes in speaking rate across an utterance. Many vocal tics resemble words or phrases uttered out of context but studies investigating the acoustic properties of tics in relation to the surrounding speech context do not exist. This state of affairs precludes understanding the relationship between linguistic vocal behavior, speech disfluencies, vocal ticking and other vocal behaviors like coughing. The current case study represents a linguistically informed analysis of vocal tic production during rehearsed, voluntary speech that investigated the impact of proximity to a prosodic boundary on tic word duration and speaking rate in an effort to determine whether vocal tics are produced via the typical speech planning and production pipeline.

3pSC11. The influence of vocal disorder on the perception of charisma in political speech. Rosario Signorello and Didier Demolin (Laboratoire de Phonétique et Phonologie, Université Sorbonne Nouvelle, Laboratoire de Phonétique et Phonologie, 19 Rue des Bernardins, Paris 75005, France, rosario.signorello@gmail.com)

Voice conveys leaders' charisma traits and triggers emotional states in listeners. A speaker with disordered voice quality is often perceived as a different individual in terms of intrinsic (personality traits and emotional states) and extrinsic characteristics (age, sex, and ethnicity) (Kreiman et Sidtis, 2011). In case of significant alteration, voice can convey significantly different charisma traits and trigger emotional states in listeners that diverge from the leaders' goals. Voice disorder can thus affect effective persuasion in the communication process. The present study reports investigations on the influence of vocal disorder on the perception of charisma traits and emotional triggering in political speech. Audio stimuli from non-disordered and disordered voice conditions of two politicians (Luiz Inácio Lula da Silva, former president of Brazil and Umberto Bossi, former leader of the Italian Lega Nord party) were collected. First, voice profiles reporting specific acoustic patterns during the two conditions were created. Secondly, cross-cultural perceptual tests of both conditions were conducted with French listeners to determine if the disordered voice condition (a) still conveys charisma and leadership traits, (b) what type of emotional states it triggers in listeners, and (c) how it influences listeners' voting behavior. [Work supported by grant DS0707-2015 ArtSpeech.]

3pSC12. Developing a remotely deliverable digit triplet in noise test for detecting high frequency hearing loss. Lina Motlagh Zadeh, Noah H. Silbert (Univ. of Cincinnati, 3239 Bishop St. Apt. #4, Cincinnati, OH 45220, motlagla@mail.uc.edu), Katherine Sternasty (Speech-Lang. Pathol. and Audiol., Miami Univ., Cincinnati, OH), and David R. Moore (Commun. Sci. Res. Ctr., Cincinnati Children's Hospital, Cincinnati, OH)

The prevalence of late-diagnosed or often unrecognized hearing loss (HL) is higher in developing countries due to the lack of access to hearing health care services. Due to the importance of hearing screening tests in early diagnosis of HL, development of remotely deliverable screening tests that can detect HL reliably, quickly, and easily provides significant benefits, specifically for underserved population. The purpose of this research is to refine the established English digit triplet test (DTT) to improve detection of high-frequency HL. The sensitivity and specificity of the DTT for detecting high frequency HL will be analyzed for low-pass filtered speech-shaped noise with three different cut-off frequencies (2 kHz, 4 kHz, and 8 kHz). The current study will also replicate previous work showing that speech reception thresholds estimated from the DTT correlate highly with listeners' pure tone average audiometry. This research should improve the accuracy of convenient, efficient tools for diagnosing HL for millions of people who have limited access to hearing health care.

3pSC13. Dysprosodie in preschool children with autism spectrum disorders. Daphne Hartzheim, Yunjung Kim, and Ariel Johnson (Commun. Sci. and Disord., Louisiana State Univ., 86 Hatcher Hall, COMD, Baton Rouge, LA 70803, dhartz4@lsu.edu)

One of the core language characteristics of children with autism spectrum disorders (ASD) have been described as monotone speech (i.e., reduced range of pitch and loudness). In this study, we investigated acoustic properties of primary prosodic cues, loudness, pitch, and speech rate. For this purpose, we analyzed speech samples from children ages 8-15 years with and without ASD while they were answering questions about social scenes. The child was asked to describe what he or she was supposed to do in a certain situation while provided a visual stimulus. The responses were video and audio recorded for later analysis. All children with ASD had verbal skills (i.e., ability to answer questions with sufficient length for acoustic analysis), even if receptive and expressive language skills was decreased for some children according to the Clinical Evaluation of Language Fundamentals-5 (CELF-5). Each child with ASD was paired with two children that were typically developing to account for natural variability in development. At the conclusion of data collection, the audio recording was analyzed for f0 variation, intensity variation, and the number of syllables produces per second using a computer software, PRAAT.

3pSC14. Daylong acoustic amplitude from the perspective of young children with and without hearing loss. Mark VanDam, Haille Heid, Stephen James (Elson S. Floyd College of Medicine, Speech & Hearing Sci., Washington State Univ., PO BOX 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Samantha Schraven, Danette Driscoll, Amy Hardie, Stacy Cahill (Hearing Oral Program of Excellence (HOPE) of Spokane, Spokane, WA), and Daniel Olds (Elson S. Floyd College of Medicine, Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Very little is known about the acoustic characteristics of the daylong auditory environment of children, especially for those children who belong to an at-risk population such as those with hearing loss. This work looks at the daylong acoustic amplitude from the auditory perspective of young children. Naturalistic audio was collected from a wearable audio recorder. Amplitude values were collected from 814 daylong recordings from children aged 1–90 months, with 318 from children who are typically developing and 496 from children with mild- to moderate hearing loss. We compared for difference by sex, hearing status, and age. Results suggest that boys' recordings had higher amplitude than girls', and that recordings of children with mild- to moderate hearing loss had higher amplitude than children who were typically developing. There were no observed sex by hearing status or sex by age interactions. For the recordings from children with hearing loss amplitude was negatively correlated with age, but for typically developing children amplitude was positively correlated with age. Results may be important for better understanding of children with hearing loss, language and speech development, automatic processing routines (such as automatic speech recognition), and intervention or therapeutic techniques for at risk populations.

3pSC15. Using real time magnetic resonance imaging to measure changes in articulatory behavior due to partial glossectomy. Maury Lander-Portnoy, Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, landerpo@usc.edu), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Real time MRI presents an exciting new method for studying speech articulation, providing the ability to capture articulatory kinematics and coordination. The current study utilizes real time MRI to document the speech of a patient who has undergone partial glossectomy at both preoperative and postoperative time points. While previous work has studied and evaluated postoperative outcomes, the comparison of individual speakers

preoperatively and postoperatively is lacking. The possibility of compensation for the pathological articulator by other articulators necessitates the ability to image multiple articulators simultaneously to observe changes in articulatory coordination. In this paper, we present a method for quantifying changes in articulatory behavior, such as compensation for pathology, between two points in time. We find no significant changes in the patient's articulatory coordination that would indicate compensation for pathology. We attribute this to the relatively high degree of speech intelligibility preserved postoperatively for this particular patient. We do observe differences in vocal tract morphology as well as changes in the variability and principle axes of movement for the tongue. The methods we present here provide a means not only for measuring individual morphological changes, but also for observing the relationship between changes in morphology and changes in articulatory behavior.

3pSC16. Vowel space in children with residual speech sound disorders. Caroline E. Spencer, Jade Clark, Sarah M. Hamilton, and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., PO Box 670379, Cincinnati, OH 45267, spenceco@mail.uc.edu)

Children with Residual Speech Sound Disorders (RSSD) are considered to have typical speech except for a few misarticulated sounds. During the course of ultrasound biofeedback therapy, parents frequently remark that their children have improved intelligibility, even if the misarticulated sound is not yet successfully remediated. One possible explanation is that therapy improves articulatory precision and expands their articulatory action space. In this study, we collected vowel space measures for 10 RSSD children pre and post 10 sessions of ultrasound therapy. In addition, we collected vowel space measures for 10 typically developing children. Preliminary results suggest a difference between populations. Across RSSD speakers pre-therapy, formant values were variable within phonemic categories; in contrast, formant values were consistent within phonemic categories in typical speakers. Further results of the comparisons between RSSD and typical children as well as for RSSD children's performance pre and post therapy will be discussed.

3pSC17. Measuring the source waveform of esophageal speakers using a two-port transfer matrix method. Andressa Beckert Otto, Andrey R. da Silva (Mech. Eng., Federal Univ. of Santa Catarina, Campus Universitário da Trindade, Florianópolis, Santa Catarina 88020-000, Brazil, andressa.beckert@hotmail.com), and Ana C. Ghirardi (Health Sci. Ctr., Federal Univ. of Santa Catarina, Florianópolis, Brazil)

The glottal waveform is an important parameter from which relevant information related to the phonation process can be extracted. A few non-invasive techniques are available on the literature in order to obtain the glottal waveform from the external measurements of a subject's voice. These techniques normally obtain the source waveform by inverse filtering the influence of the subject's vocal tract and the mouth radiation impedance based on linear prediction algorithms. Nevertheless, such techniques are not entirely adequate for subjects with esophageal and tracheoesophageal phonation, mainly due to the aperiodic behavior of these types of waves. The present work proposes a new technique, in which the transfer matrix of a subject's vocal tract is obtained by the layer-peeling algorithm. Thereafter, the source waveform is obtained by resolving a two-port transfer matrix equation, where the source waveform and the measured voice are the input and output ports, respectively. Preliminary results using the new technique are compared with those obtained by the traditional methods involving inverse filtering and the Sondhi tube.

3pSC18. Recovering prosody of a case of foreign accent syndrome (FAS). Grace Kuo (Dept. of Classics, Modern Lang. and Linguist, Concordia Univ., 3125 Campbell Hall, Los Angeles, CA 90095, grace.kuo@concordia.ca)

Foreign Accent Syndrome (FAS) is a rare disorder characterized by the emergence of a perceived foreign accent following brain damage. In this case study, acoustic analyses were performed on the speech of a Mandarin-speaking female FAS patient at her four doctor visits. The reading materials included news in newspaper and a tongue twister. The acoustic analyses include sentence-level intonation and rhythm measures such as %V and PVIs. Results reveal a gradual recovery trajectory from a disfluent stressed-timed pattern to a fluent syllable-timed pattern. A heritage Mandarin speaker and an advanced nonnative speaker recorded the same reading materials and the same acoustic analyses were performed on their speech for comparison.

3pSC19. Phoneme production by Hispanic hearing-impaired children. Tanya Flores (World Lang. and Cultures, Univ. of Utah, 255 S Central Campus Dr., LNCO 1400, Salt Lake City, UT 84109, Tanya.Flores@utah.edu)

This study examines the speech productions of Hispanic deaf and hearing impaired (DHI) children from 3 to 7 years of age who have had delayed medical intervention. The data presented here is from the initial data collection session and will focus on the segmental phonemic productions of the target group as compared to the productions of the control groups (Hispanic peers with normal hearing and non-Hispanic DHI peers) from the same local community. The goal of the study is to create a speech corpus of Hispanic DHI children that will be used to study various aspects of their language development. Findings will contribute to the currently limited acoustic research on minority DHI children whose home language differs from their specialized language program (in this case, English Listening and Spoken Language Program).

Session 3pSP

Signal Processing in Acoustics and Underwater Acoustics: Detection, Classification, Localization, and Tracking (DCLT) Using Acoustics (and Perhaps Other Sensing Modalities) IV

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

Invited Papers

1:00

3pSP1. The Virtual Ocean—A high-fidelity, physics-based testbed for distributed, autonomous undersea acoustic sensing networks. Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-204, Cambridge, MA 02139, henrik@mit.edu)

Behavior-based autonomy enables unmanned vehicles to adapt to the current environmental and tactical situation, with behaviors individually optimized for the detection, classification, localization, and tracking phases of the sensing mission. However, the adaptation has limited predictability with platform actions potentially becoming suboptimal or risky, in turn requiring extensive experimentation and testing. For obvious cost reasons, simulations are critical to this process. However, to be reliable, it is critical to the robustness evaluation that the processing chain and autonomy system is identical to the one operated on the physical platforms, with only the sensor stimulation and the platform dynamics being simulated. The Virtual Ocean is a physics-based ocean environment testbed, developed for carrying out virtual experiments with MOOS-ivP platform autonomy systems for environmentally and tactically adaptive acoustic sensing and communication networks. Coupled to ocean modeling frameworks such as MSEAS and HYCOM, and legacy acoustic models, it supports virtual experiments involving multiple fixed or mobile nodes with arbitrary volumetric or towed arrays, operating in realistic 3D ocean environments. Examples and experimental validations will be discussed. [Work supported by ONR and DARPA.]

1:20

3pSP2. Multitouch ultrasonic touchscreen. Kamyar Firouzi and Butrus T. Khuri-Yakub (Ginzton Lab, Stanford Univ., 348 Via Puebla Mall, Rm. 102, Stanford, CA 94305, KFIROUZI@STANFORD.EDU)

Touchscreen sensors are widely used in many devices such as smart phones, tablets, laptops, etc. We present the design, analysis, and implementation of an ultrasonic touchscreen system that utilizes interaction of transient Lamb waves with objects in contact with the screen. The governing principle revolves around the propagation and reverberation of guided elastic waves in a bounded space, in which the localization of touch points can be challenging. Reverberant fields in enclosures can potentially carry useful information, however, in an incoherent way. Incoherency comes from consecutive reflections of the wave energy several times in the domain, ultimately leading to mixing of the wave energy in a seemingly random way. However, spreading of the wave energy can lead to multiple interrogations of each point in the enclosure. Hence, temporal information buries information about substructural changes making it feasible to conduct only a few spatial measurements. We present a learning localization algorithm capable of localizing multiple simultaneous touch points (up to 12 touches on a tablet) and with a very limited number of measurements (one or two). This in turn can significantly reduce the manufacturing cost. We also present an algorithm to improve on robustness to environmental and thermal noise.

Contributed Papers

1:40

3pSP3. Dispersion correction for acoustic borehole logging data. Said Assous and Peter Elkington (GeoSci., Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com)

Compressional and shear formation velocities are key to the prediction of petrophysical properties from seismic attributes. In fast formations shear velocity may be obtained from monopole source, but in slow formations, it is commonly determined from the flexural mode associated with dipole excitation, which is a dispersive borehole-guided mode who's low frequency and high frequency asymptote to the formation S-velocity, and to the Scholte-velocity, respectively. The shear slowness is commonly computed from well log dipole flexural mode data using Semblance Time Coherence (STC) processing. Dispersion is handled by restricting the waveforms spectral content to the low frequencies that travel close to the formation's shear velocity. This restricting may not eliminate the need for a residual dispersion correction. Inversion addresses this difficulty by computing shear slowness directly from observed dispersion characteristics. In order to make the inversion efficient the iterative steps which compare observed and forward modeled dispersion curves are replaced with a neural net trained on a large number of pre-modeled curves generated with known formation and borehole properties. Automated mode frequency detection constrains the bandwidth over which dispersion curves are matched. Results from 127,000 modelled and field data points show improved accuracy and precision relative to STC processing.

1:55

3pSP4. Damage localization based on the reconstructed Green's function from a diffuse noise field on a thin rectangular aluminum plate. Sun Ah Jung (Seoul National Univ., Bldg. 36 212, 1 Gwanak-ro, Gwanak-gu, Seoul 151-742, South Korea, sunj@snu.ac.kr), Keunhwa Lee (Sejong Univ., Seoul, South Korea), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

The extracted Green's function from ambient noise cross-correlation is applied to localize a damage on an aluminum plate using a passive sensor array. Damage was investigated by drilling holes at various locations with different sizes. The localization process uses a subtracted Green's function of an undamaged plate from that of a damaged plate, which can reveal the scattered wave from the damage boundary. Localization algorithms used in active structural health monitoring, including the time of arrival, the time difference of arrival, the energy arrival and the Rayleigh maximum likelihood estimation [Flynn *et al*., Proc. R. Soc. Lond. A. Math. Phys. Sci. (2011)] methods are applied to the subtracted signal according to the number of sensors and that of noise sources. The subtracted signals are band pass filtered with various center frequency resulting in multiple images which are used in image fusion to increase the accuracy of localization [Michaels *et al*., Wave motion. (2007)]. The performance of the algorithms demonstrates the efficacy of the passive sensing-based localization method even when using small number of sensors and sources.

2:10

3pSP5. Parameter estimation of acoustic scattering echoes of underwater targets using sparse representation approach. Xiangxia Meng, Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, Heilongjiang 150001, China, mengxiangxia@hrbeu.edu.cn), and Andreas Jakobsson (Dept. of Mathematical Statistics, Lund Univ., Lund, Sweden)

The acoustic scattering echo of an underwater target usually consists of multiple overlapped components. The time delay of each reflection is of great importance to estimate the size of the target. Herein, a sparse approach is presented to obtain the estimation of time delays with high resolution and accuracy. The parameters are determined using a dictionary with many more elements than expected reflections, with each dictionary element being found as the convolution of the transmitted signal and a potential impulse. The estimation problem is then converted into a convex optimization problem, which is then solved efficiently using the alternating direction method of multipliers (ADMM) framework. The estimation accuracy depends on the grid of the dictionary. To obtain high resolution, a dictionary refinement technique is employed. To model the time-varying nature of the signal amplitudes, we further estimate the temporal envelope of the signal using a weighted combination of splines. Using this technique, the algorithm can estimate both the time-delay and amplitude of a reflection simultaneously, without the need of prior information about the number of components. The method can be applied for the separation of overlapped components and high-resolution estimation of multi-angle echoes.

2:25

3pSP6. Scattered signal distributions, parametric uncertainties, and Bayesian sequential updating. D. K. Wilson, Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), Daniel J. Breton (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Edward T. Nykaza (Construction Engineering Res. Laboratory, U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Vladimir E. Ostashev (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A variety of probability density functions (pdfs) have been proposed for scattered signals, which have varying analytical advantages and ranges of physical applicability. We discuss here situations modeled by a compound pdf, in which a basic pdf, attributable to the underlying scattering process, has uncertain parameters or is modulated by variability in the environment. The parameters of the modulating pdf are termed *hyperparameters*. Some previous examples of compound formulations include the K-distribution, for which strong scattering (exponential pdf) is modulated by a gamma pdf for the mean signal power, and scattering by intermittent turbulence, for which strong scattering is modulated by a log-normal pdf for the structure-function parameter. We describe some alternative formulations, including strong scattering modulated by a gamma pdf for the *inverse* mean power, and Rytov (log-normal) scattering modulated by a normal pdf for the log-mean of the signal. These lead to relatively simple marginalized signal power distributions (Lomax and log-normal, respectively). Furthermore, the conditional scattered signal pdf may be viewed as a likelihood function in which the modulating pdf is the Bayesian conjugate prior. Hence the hyperparameters of the modulating process can be refined by simple sequential Bayesian updating as additional transmission data become available.

2:40

3pSP7. Implications of the evolving sound speed field on detection and classification of targets in shallow water. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Alan Blumberg (Stevens Inst. of Technol., Hoboken, NJ), Peter J. Stein (Sci. Solutions Inc., Nashua, NH), Eric I. Thorsos (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA), and Amy Gruhl (Sci. Solutions Inc., San Diego, CA)

Acoustic refraction caused by gradients in the sound speed field within a region covered by an active sonar system affects its performance and quality of information. From the perspective of target strength, refraction can focus sound and enhance target returns, or it can redirect sound and create shadow zones where targets will drop out. These effects depend on the evolving sound speed field, as well as the placement of the transmitters and receivers. In addition, focused sound at the water-sediment interface can produce a small region of enhanced backscatter, which can rise above the nominal reverb level by 6 dB or more. Continuous evolution of the time-dependent sound speed field can cause these relatively compact returns to appear as moving targets, increasing false-alarm detections. Advances in fine scale oceanographic models facilitate studying these propagation effects, both the enhanced/degraded target returns and the target-like bottom backscattered features. A combined acoustic and oceanographic model is presented for discussion on what conditions cause these effects, the possibility of identifying their occurrence within data, and what the consequences are on a practical sonar system operating in shallow water.

2:55

3pSP8. Environmental calibration curves. Gerald L. D'Spain, Dennis Rimington, Shih-Hsuan (Shane) Yuan (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), and Tyler A. Helble (SPAWAR Systems Ctr. Pacific, San Diego, CA)

Sound propagation through the ocean causes distortion of the received waveform, characterized by the impulse response of the channel (which often assumes a fixed medium since ocean Mach numbers are so small and the temporal scales of oceanographic variability typically are much larger than signal travel times). In addition, noise, both its level and temporal character, impact signal detection and processing. The purpose of this presentation is to describe a physics-based approach to developing environmental calibration curves to account for environmental effects in the received field. The methods are first applied to passive acoustic recordings by fixed, single-hydrophone packages deployed in the Southern California Bight. Five types of baleen whale calls from blue, fin, and humpback whales are the signals of interest. Results show that these environmental calibration curves, determined by the probability of detection, are most sensitive to sediment thickness over the blue and fin whale frequency band, whereas they are most sensitive to sediment type for humpback calls. The statistical framework for estimating the mean, bias, and variance of these environmental calibration curves also is discussed. [Research supported by the Living Marine Resources Program, the Office of Naval Research, and the Joint Industry Programme.]

3:10

3pSP9. Estimation and equalization for shallow water communication channel using geometric encoding of channel multipath. Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

The shallow water acoustic channel is well-known to exhibit rapid fluctuations in its impulse response, which is challenging to estimate and compensate for in real time. In particular, the moving ocean surface and static sea bottom reflect the acoustic signal in unpredictable ways to create rapidly time-varying multipath arrivals. The talk will focus on the nexus of shallow water acoustic propagation paths and how multipath can be discovered and exploited to improve shallow water acoustic communications. In particular, the talk will review sampling strategies for the time-varying shallow water acoustic channel and propose novel geometric encoding techniques that exploit non-uniform channel sampling strategies. We will also demonstrate how geometric encoding can be harnessed for real-time channel estimation as well as semi-blind channel equalization. Results based on experimental field data in recent work as well simulation results will be presented.

Plenary Session and Awards Ceremony

Marcia J. Isakson,
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New ASA Fellows

John S. Allen, III – For contributions to the understanding of ultrasound contrast agents

Kelly Benoit-Bird – For contributions to marine ecological acoustics

John J. Loverde – For contributions to quantification and understanding of building response to sound and impact

Alexander Ya Supin – For contributions to the understanding of auditory systems of humans and marine mammals

Introduction of Scholarship and Award Recipients

Curtis Wiederhold, recipient of the 2017 Frank and Virginia Winker Memorial Scholarship
for Graduate Study in Acoustics

Matthew Neal, recipient of the 2017
Leo and Gabriella Beranek Scholarship in Architectural Acoustics and Noise Control

Introduction of Andone C. Lavery, recipient of the 2017 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

Ryan Kellman, recipient of the Science Writing Award in Acoustics for Journalists for his video
Singing Ice: A Star Wars Story (NPR.org, 2016)

Tyler Adams, recipient of the Science Writing Award for Professionals in Acoustics for
Sound Materials: A Compendium of Sound Absorbing Materials for Architecture and Design
(Frame Publishers, Amsterdam, 2016)

David T. Bradley, Erica E. Ryherd, and Lauren M. Ronsse, *recipient of the Science Writing Award for Professionals in Acoustics* for their book

Worship Space Acoustics: 3 Decades of Design (Springer Science+Business Media, New York (ASA Press), 2016)

Robert D. Celmer, recipient of the 2015 Rossing Prize in Acoustics Education

Michael J Buckingham, Pioneers of Underwater Acoustics Medal

Evgenia Zabolotskaya, Silver Medal in Physical Acoustics

David Griesinger, Wallace Clement Sabine Medal

Session 3eED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

This workshop for New Orleans area Girl Scouts (age 12–17) consists of hands-on tutorials, interactive demonstrations, and discussion about careers in acoustics. The primary goals of this workshop are to expose girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please e-mail Keeta Jones (kjones@acousticalsociety.org) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. to 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. to 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY EVENING, 6 DECEMBER 2017 BALCONY I/J/K, 5:00 P.M. TO 8:00 P.M.

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, 5 December

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Studio 7
Acoustical Oceanography	7:30 p.m.	Salon A/B/C
Animal Bioacoustics	7:30 p.m.	Salon F/G/H
Architectural Acoustics	7:30 p.m.	Studio 9
Musical Acoustics	7:30 p.m.	Studio 4
Physical Acoustics	7:30 p.m.	Balcony L
Psychological and Physiological Acoustics	7:30 p.m.	Balcony M
Structural Acoustics and Vibration	8:00 p.m.	Studio 7

Committees meeting on Wednesday, 6 December

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Balcony M
Signal Processing in Acoustics	7:30 p.m.	Salon D

Committees meeting on Thursday, 7 December

Committee	Start Time	Room
Noise	7:30 p.m.	Studio 2
Speech Communication	7:30 p.m.	Salon A/B/C
Underwater Acoustics	7:30 p.m.	Salon F/G/H

ACOUSTICAL SOCIETY OF AMERICA
PIONEERS OF UNDERWATER ACOUSTICS MEDAL

Michael J. Buckingham
2017

The Pioneers of Underwater Acoustics Medal is presented to an individual irrespective of nationality, age, or society affiliation, who has made an outstanding contribution to the science of underwater acoustics, as evidenced by publication of research in professional journals or by other accomplishments in the field. The award was named in honor of five pioneers in the field: H. J. W. Fay, R. A. Fessenden. H. C. Hayes, G. W. Pierce, and P. Langevin.

PREVIOUS RECIPIENTS

Harvey C. Hayes	1959	Ivan Tolstoy	1990
Albert B. Wood	1961	Homer P. Bucker	1993
J. Warren Horton	1963	William A. Kuperman	1995
Frederick V. Hunt	1965	Darrell R. Jackson	2000
Harold L. Saxton	1970	Frederick D. Tappert	2002
Carl Eckart	1973	Henrik Schmidt	2005
Claude W. Horton, Sr.	1980	William M. Carey	2007
Arthur O. Williams	1982	George V. Frisk	2010
Fred N. Spiess	1985	Michael B. Porter	2014
Robert J. Urick	1988		

CITATION FOR MICHAEL J. BUCKINGHAM

". . . for contributions to the understanding of ocean ambient noise and marine sediment acoustics."

NEW ORLEANS, LOUISIANA • 6 DECEMBER 2017

After a few words in conversation with Mike, it isn't difficult to figure out where his roots are, the accent gives it away. Mike Buckingham grew up in the suburban North London district of Edgware, about 20 km northwest of Big Ben. Very early on as a young student, Mike developed a consuming interest in math and physics, so it followed naturally that he enrolled in graduate school in science at the nearby University of Reading in 1967. Like many others of that era, Mike's route into underwater acoustics was not direct, but through another field of research, in his case, solid state physics. It's not clear if he had any idea as a graduate student that his research career would be focused on sound in the ocean, but there was one aspect of his PhD studies that translated smoothly into his future work. His thesis research involved the study of noise, electronic noise in silicon p-n junctions, work that subsequently became the basis of a book, "Noise in Electronic Devices and Systems," published a few years after his graduation in 1971. While working on the book, Mike had already moved on to defense research in the Royal Aerospace Establishment, where he spent a lot of time flying in Royal Air Force BAC 1-111 twin-engine jet aircraft flown by Royal Air Force test pilots on missions to deploy hydrophones in the Arctic Ocean. He quickly realized that the ocean is a very noisy environment, probably a lot noisier than a p-n junction. His first papers on ocean noise and sound propagation earned early recognition of the A.B. Wood Medal of the Institute of Acoustics in 1982, and attracted considerable attention on this side of the Atlantic. Mike was soon in high demand as a visiting scientist in the United States. After a number of research appointments that kept a foot on each side of the Atlantic, he and his charming wife, Penny, settled in San Diego in 1990 where he is currently Distinguished Professor in the Marine Physical Laboratory at the Scripps Institute of Oceanography.

The central theme of Mike Buckingham's research is ambient noise in the ocean. His research into the physical nature of ambient noise, both experimentally and in theoretical model development, has created and stimulated the active and highly productive research field of ambient noise oceanography. Before Mike's pioneering insight that was published in 1987, ambient noise in the ocean was viewed as a problem, at the very least a nuisance that corrupted or masked the signals we were trying to detect. Now acousticians the world over use ambient noise as a valuable signal in its own right. Present day research using ambient noise as natural sound sources for applications such as geoacoustic inversion and imaging of objects in the ocean owes its existence to Mike's work. The timeliness of this approach cannot be overstated given the general concern about impacts of anthropogenic generated sound on marine life.

Perhaps the most publicized of Mike's work in underwater acoustics was his research on acoustic daylight, the use of ambient noise for imaging objects in the ocean. This work was a showpiece for underwater acoustics that captured widespread attention scientifically (a publication in *Nature* in 1992) and in the popular journals (an article in *Scientific American* in 1996).

His experience with ambient noise for geoacoustic inversion introduced him to a challenging research problem in marine sediment acoustics: dispersion of sound in sediments. Mike questioned the conventional ideas about sound propagation in porous marine sediments, and announced a new approach, the grain-shearing theory, published in 1997. It challenged the accepted approach for sound propagation in porous sediment media developed by Maurice Biot over 60 years ago, and made an immediate impact in geoacoustic research, stimulating new experimental and theoretical research, for example, the Office of Naval Research Sediment Acoustics Experiments in 1999 and 2004, SAX'99 and SAX'04. Mike's continued development of his viscous grain-shearing theory consolidates his place as a central figure in marine sediment acoustics.

Mike is an experienced aircraft pilot and is passionate about flying, to the extent that it eclipses his interest in and commitment to most other events, including plenary sessions of the Acoustical Society of America. I have to confess being corrupted by Mike to skip out for a Wednesday afternoon aerial tour of San Diego while the rest of the society was in plenary session during the meeting held in that city a few years ago. But flying is not all fun and games for Mike, he turned it into serious business, proposing that aircraft noise could be a useful sound source for characterizing geoacoustic properties of the ocean bottom. He worked out the theory for coupling the aircraft sound into the ocean bottom, and then demonstrated the concept by flying his aircraft as the sound source in a simple but sensational experiment along the San Diego coastline within full view of the Director's office at Scripps. Fortunately, the flight time was not charged against annual leave.

Most recently Mike has focused his attention on another extreme, measurement of ambient noise in the deep ocean, that is, at the deepest known depths. Mike, along with his graduate students, designed, built and successfully used a unique instrument to record ambient noise as it descended to the deepest depths in the Mariana Trench. The deployment of the instrument established a record for depth of noise measurement when it dropped into the Challenger Deep.

Throughout his research career, Mike has challenged conventional ideas in underwater acoustics and provided fresh insight that has led us in new directions that are recognized as paradigm shifts in our research. His trademark approach is characterized by pulling together ingenious and bold theoretical advances with clever and simple experiments, all carried out with flair and elegance at the same time. He has published over 40 papers in the *Journal of the Acoustical Society of America*, each one of which serves as an excellent example for young researchers in clarity and logic in scientific writing. His leadership as a founding member and inaugural Chair of the Technical Committee on Acoustical Oceanography and his unselfish service in many other roles has enriched the Acoustical Society. It is a great pleasure to introduce Mike Buckingham, avid pilot, renowned acoustical oceanographer and truly, Pioneer in Underwater Acoustics.

N. ROSS CHAPMAN

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Physical Acoustics

Evgenia Zabolotskaya

2017

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

Isadore Rudnick	1975	Robert E. Apfel	1997
Martin Greenspan	1977	Gregory W. Swift	2000
Herbert J. McSkimin	1979	Philip L. Marston	2003
David T. Blackstock	1985	Henry E. Bass	2006
Mack A. Breazeale	1988	Peter J. Westervelt	2008
Allan D. Pierce	1991	Andrea Prosperetti	2012
Julian D. Maynard	1994		

CITATION FOR EVGENIA ANDREEVNA ZABOLOTSKAYA

". . . for contributions to nonlinear acoustics and bubble dynamics"

NEW ORLEANS, LOUISIANA • 6 DECEMBER 2017

Evgenia (Zhenia) Andreevna Zabolotskaya was born and raised in Moscow. Her grandfather worked as a janitor in the Kremlin and lived there with his family before and after the Russian Revolution, and during the 1930s the Soviet secret police took her mother away from their home for several days because she was active in her church. Zhenia demonstrated a proclivity for math and physics at a young age, so it was no surprise that she was admitted to the prestigious Physics Department at Moscow State University (MSU), where some of her classes were taught by none other than Lev Landau, who received the Nobel Prize in Physics in 1962. She still recalls Landau's intimidating challenge on the first day his electrodynamics class, "If you do not understand quantum mechanics, you should think about whether you have chosen the right path in life."

While at MSU, Zhenia met Yurii (Yura) Ilinskii, a remarkable young theoretical physicist. After Zhenia and Yura graduated, she with her bachelor's degree and he with his PhD, both were employed at the Electro-Mechanics Institute in Moscow, where they started dating, and in 1963 they married. That same year Zhenia returned to MSU to pursue a PhD in the Wave Processes Chair under Rem Khokhlov in the Physics Department. By the 1970s, the work conducted in Khokhlov's chair made MSU a leading international center for theoretical nonlinear acoustics and, even more influentially, for nonlinear optics.

Zhenia was among the first graduate students in Khokhlov's nonlinear acoustics group. Other members of the group at that time included Stepan Soluyan, Oleg Rudenko, Vyacheslav Kuznetsov, and Nellie Pushkina. One of Zhenia's four papers that culminated in her PhD in 1968 contained nothing less than a model equation for nonlinear bubble dynamics that became widely used in the former Soviet Union instead of the Rayleigh-Plesset equation that was used extensively in the West [Sov. Phys. Acoust. 13, 254 (1967)]. The same paper also contained the first effective medium theory for nonlinear propagation of sound in bubbly liquid.

In 1966, while still a doctoral student, Zhenia joined the Acoustics Institute in Moscow, sharing an office with other Khokhlov protégés Anna Polyakova and Konstantin Naugolnykh. After receiving her PhD, her continued collaboration with Khokhlov produced one of the most famous advances in twentieth-century nonlinear acoustics: the Khokhlov-Zabolotskaya equation for nonlinear sound beams [Sov. Phys. Acoust. 15, 35 (1969)]. The KZ equation became known as the KZK equation after Kuznetsov added a term accounting for losses in 1970.

The impact of the KZK equation on practical applications of nonlinear acoustics, whether biomedical, industrial, or sonar related, cannot be overstated. Originally important for describing nonlinear effects in sonar (1970s and 80s), for the past two decades the KZK equation has provided the main theoretical basis for modeling HIFU (High Intensity Focused Ultrasound), a therapeutic procedure for treating cancer. Reports of modeling HIFU using the KZK equation are now presented routinely at ASA meetings. Subsequent work by Zhenia with mathematicians Nikolai Bakhvalov and Yakov Zhileikin was published in 1982 in their book *Nonlinear Theory of Sound Beams*, which was deemed so groundbreaking that it was translated into English by Robert Beyer for the American Institute of Physics.

Zhenia returned to MSU in 1971 where she was appointed not in the Physics Department but, interestingly, in the Biology Department, and in 1982 she joined the General Physics Institute of the USSR Academy of Sciences. In 1985 she was awarded the USSR State Prize for her overall contributions to nonlinear acoustics, a level of recognition that today is bestowed in the Kremlin and accompanied by personal congratulations from the Russian president.

Zhenia's path to the United States began unwittingly in 1982 in Tallinn, Estonia, where she encountered David Blackstock from the University of Texas at Austin (UT), basically

Khokhlov's American counterpart in nonlinear acoustics, at a symposium on nonlinear deformation waves. Blackstock maintained contact with Zhenia following that symposium, and in 1987 he encouraged her to meet his UT faculty colleague and former graduate student Mark Hamilton at the International Symposium on Nonlinear Acoustics to be held that August in Novosibirsk, in the heart of Siberia. Zhenia and Mark hit it off, because his research at the time was based largely on the KZK equation, and in 1988 he returned to the USSR specifically to visit her.

Zhenia then visited Hamilton in the Mechanical Engineering Department at UT for several months in both 1989 and 1990, and in 1991 she moved to Austin with Yura and their two daughters. She immediately published a theoretical model for nonlinear Rayleigh waves [J. Acoust. Soc. Am. 91, 2569 (1992)], a type of interface wave in solids, initiating a line of research that continued for over a decade. In collaboration with UT students, her fundamental theory was extended to include Scholte and Stoneley waves, and also surface waves in anisotropic and piezoelectric materials. Nonlinear interface waves are generated on very large scales by earthquakes and on very small scales by piezoelectric actuators in microfluidic and SAW (Surface Acoustic Wave) devices.

In 1997 Zhenia and Yura took a position at a startup company in Virginia at which two former UT doctoral students hired them to develop theoretical models for nonlinear sound fields in resonators. Zhenia and Yura returned to UT's Mechanical Engineering Department in 2000 to work on nonlinear phenomena in thermoacoustic engines. Then in 2003 UT's Applied Research Laboratories hired Zhenia and Yura to initiate a research program in biomedical acoustics. This appointment enabled Zhenia to return to one of her original research interests, nonlinear bubble dynamics, but this time for medical applications such as shock-wave lithotripsy.

Zhenia and Yura retired from their research staff positions at UT in February 2015. They continue to live in Austin where they remain engaged in acoustics research at UT, as well as consult for a startup company in Massachusetts for which they model particle manipulation with acoustic radiation force.

Zhenia's contributions to theoretical nonlinear acoustics are prolific and foundational. Her publications are not only exclusively on nonlinear acoustics but equally divided between *Soviet Physics-Acoustics* in the first half of her career and the *Journal of the Acoustical Society of America* in the second half. The name Zabolotskaya is justifiably iconic in the history of nonlinear acoustics.

MARK F. HAMILTON
OLEG A. SAPOZHNIKOV

WALLACE CLEMENT SABINE AWARD
OF THE
ACOUSTICAL SOCIETY OF AMERICA

David Griesinger

2017

The Wallace Clement Sabine Award is presented to an individual of any nationality who has furthered the knowledge of architectural acoustics, as evidenced by contributions to professional journals and periodicals or by other accomplishments in the field of architectural acoustics.

PREVIOUS RECIPIENTS

Vern O. Knudsen	1957	Richard V. Waterhouse	1990
Floyd R. Watson	1959	A. Harold Marshall	1995
Leo L. Beranek	1961	Russell Johnson	1997
Erwin Meyer	1964	Alfred C. C. Warnock	2002
Hale J. Sabine	1968	William J. Cavanaugh	2006
Lothar W. Cremer	1974	John S. Bradley	2008
Cyril M. Harris	1979	J. Christopher Jaffe	2011
Thomas D. Northwood	1982	Ning Xiang	2014

SILVER MEDAL IN ARCHITECTURAL ACOUSTICS

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 RECIPIENT

Theodore J. Schultz 1976

CITATION FOR DAVID GRIESINGER

". . . for contributions to the understanding of electroacoustics and human perception of sound"

NEW ORLEANS, LOUISIANA • 6 DECEMBER 2017

David Griesinger was born in Cleveland Ohio. He attended Harvard University, earning a B.A. degree in 1966, M.A. degree in 1968 and a PhD in physics in 1976. He is the recipient of the Gold Medal of the Verband Deutscher Tonmeister, Silver Medal of the Audio Engineering Society, and the Peter Barnett Award from the Institute of Acoustics. While attending graduate school, he met his wife Harriet who is also a physicist. Together, they have one son Ben, and one grandchild.

David's father was a versatile musician. He had one of the first FM radios, as well as a large collection of 78 rpm records. He would often organize chamber music evenings at their home with a trio, quartet, or quintet. In the early 1950's David's father purchased the first consumer magnetic tape recorder made in America, and David soon learned to use it. Several years later David convinced his mother to purchase "Magnetic Tape Recording" by Marvin Cameras in lieu of a book from the children's section of the bookstore. In less than two years he had built his own tape machine "out of old motors, ball bearings, rubber belts, vacuum tubes, and wood." He was also singing in the church choir, the school glee club and learning to play French horn.

While attending Harvard, David recorded every concert that he could, working with many notable conductors and musicians. He modified tape recorders, and built his own electronics to improve sonic quality, including an 8-channel mixer with independent VU meters, which was a novelty in its day. He even machined and built his own condenser microphones which he states "were a vast improvement" over the dynamic microphones he was using previously. A source of frustration was the inability to correct balance between instruments while also optimizing their balance in the reverberation. Making this correction was time consuming, tedious, and subject to interruption.

By this time David had built his own microcomputer, and the price of memory had dropped sufficiently to contemplate building a digital reverb. After constructing several prototypes, he had a system that produced the desired results, which became the basis for the Lexicon 224 digital reverb. The 224 and its successors (224X and 224XL) created a paradigm shift in the audio recording industry. It delivered unprecedented acoustic flexibility in a chassis that was smaller than a bread box. Its portability made it accessible to any studio, producer, or engineer as a rent in for specific sessions, or live recordings. In addition, it enabled artists to specify the system for touring productions.

The 224 was quickly recognized as the gold standard of its day. However, the audio community was about to undergo another paradigm shift in the form of digital media. This spawned the development of the next generation of the 224–the Lexicon 480L. The 480L was the platform that allowed David to create both a new digital reverb as well as audio tools in the digital domain. This included one of the first digital mixers, digital compression, and digital EQ. Many other tools followed. David built a Dolby compatible 4:2:4 surround decoder that automatically corrected azimuth errors in VHS recordings. The resulting product, the Lexicon CP-1, started Lexicon's consumer products division. Subsequent work led to a patent for the Logic 7 surround sound process. Throughout this time, David remained active with recording and acoustics research, publishing numerous papers on subjects ranging from microphone technique to binaural recording surround sound and acoustics.

In the late 1980's, Neil Muncy asked if David had any experience using digital reverberation to alter physical spaces. Neil had been hired by the Ontario Heritage Foundation to oversee the acoustics of the Elgin Theatre, one of the last remaining Lowes double decker venues (theatre built on top of a theatre) in the world. Past attempts at electronic acoustic augmentation had produced systems with great complexity, limited flexibility, and less than optimum sonic quality. David decided to investigate what was necessary to create a system that was both practical, and capable of making a substantial improvement in acoustic quality. This research led to the development of LARES (Lexicon Acoustic Reinforcement and Enhancement System) and a patent for the system. The first system was installed in the Elgin Theatre, and since that time hundreds of similar systems have been installed throughout the world to high critical acclaim.

Working with LARES afforded David exposure to both a wide range of venues as well as many practitioners working in architectural acoustics. It also provided the opportunity to work with numerous conductors and musicians. His focus began to shift to the physics involved in the perception of reverberation and the neural processing of human hearing. This research has led to a series of discoveries conveyed in papers starting in 2004 ("Pitch Coherence as a Measure of Apparent Distance and Sound Quality in Performance Spaces" IOA 2006; "Phase Coherence as a Measure of Acoustic Quality, Part One: The Neural Network." ICA 2010). By 2010, David had described the neural mechanisms that invoke the sensations of acoustic distance and acoustic intimacy or "engagement," and had developed the means to reliably measure it, which he dubbed LOC (The measure LOC predicts the threshold for localizing speech in a diffuse reverberant field, based on the strength of the direct sound relative to the build-up of reflections in a 100 ms window.). These mechanisms also explain many other aspects of aural perception including the cocktail party effect, the ability to tune instruments to high precision, the reasons we hear pitch in octaves, why fifths and fourths sound so harmonious, why we are so good at hearing signals buried in noise, why mechanical speech to text systems are >10 dB worse than human hearing in the presence of noise, why we can separate two simultaneous talkers into two independent neural streams, if they are different in pitch by just over ¼ semitone, and why children can't remember what teachers say in most classrooms. This research also resulted in new algorithms for electronic acoustic enhancement that provide a substantial improvement in perceived clarity.

The implications of this research for acoustic design of spaces built for music and speech is substantial. It represents an equivalent paradigm shift in the field of architectural acoustics to similar paradigm shifts that David has instigated throughout his career. His enduring interest in the human perception of sound is manifested by the ongoing research, continued writing and publishing of technical papers, and true inventions in the field to which he has contributed so much.

STEPHEN BARBAR

Session 4aAA

Architectural Acoustics: Speech Intelligibility in Reverberation and Noise

Roger W. Schwenke, Chair
Meyer Sound Laboratories, 2832 San Pablo Ave., Berkeley, CA 94702

Chair's Introduction—8:25

Invited Papers

8:30

4aAA1. Speech intelligibility studies in a historic multipurpose room. Ana M. Jaramillo (Olson Sound Design, LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu), Peggy B. Nelson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Bruce C. Olson (AFMG Services North America, LLC, Brooklyn Park, MN)

The Speech-Language-Hearing Sciences department at the University of Minnesota is located in Shevlin Hall, a historic building dating from the 1920s. Shevlin Hall has a large room on the first floor with a high coffered ceiling that is often used for receptions and presentations but also serves as a classroom. Students with hearing loss often complained about attending classes in Shevlin 110 due to its high reverberation times, combined with window AC unit noise, resulting in very degraded speech intelligibility. In 2013, the room went through renovations that addressed room acoustics, sound system design, as well as lighting and equipment (not including AC). Post-renovation measurements and subjective impressions have shown a significant improvement in speech intelligibility in the room. For this paper, we are looking at subject-based intelligibility tests to compare with prediction metrics and simulations, as well as collecting a qualitative narrative on the department's impression on the room before and after renovations.

8:55

4aAA2. A comparison of speech coherence index and measured speech intelligibility. Tobi A. Szuts and Roger W. Schwenke (Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94610, tobi@meyersound.com)

Perceptual tests were taken in real and simulated spaces to determine whether Speech Coherence Index (SCI) predicts human performance better than Speech Transmission Index (STI). SCI is a proposed method of estimating speech intelligibility during event conditions, that is, in real time with program material. The complex valued coherence function is used to estimate the signal-to-noise ratio on a per frequency basis, and is calculated with short time windows at high frequencies and longer time windows at low frequencies to mimic the multi-resolution nature of human hearing. SCI has been shown to track STI very closely under simplified conditions. Under more realistic conditions, SCI is more sensitive to reverberant energy and is always less than STI. Under certain conditions, such as long source to listener distance or low SNR, SCI predicts significantly lower intelligibility values than STI.

9:20

4aAA3. Acoustical renovation and sound system design in a church in buenos aires, argentina. Fernando M. del Solar Dorrego (Elec. Eng., Instituto Tecnológico de Buenos Aires, 445 Waupelani Dr., Apt. B10, State College, PA 16801, fsolar@gmail.com) and Pablo Gardella (Elec. Eng., Instituto Tecnológico de Buenos Aires, Buenos Aires, Ciudad de Buenos Aires, Argentina)

Church "Nuestra Señora de la Paz" suffered from an excessive reverberation (4 seconds at mid-frequencies) and an ill-suited sound reinforcement system. Measurements of the Speech Transmission Index (STI) in the congregation area indicated that speech intelligibility was greatly impaired. Thermal insulation on the roof was insufficient and impacted in air conditioning costs. Renovations included the design of a sound-absorbing and thermally insulating covering in the roof area. Subsequent measurement of room acoustic parameters showed that the proposed goals were met. A new sound system, which increased speech intelligibility and audio quality substantially, was designed and tested. The possible appearance of acoustical defects after the renovation, such as echoes and flutter echoes, was studied using objective tests on the measured Room Impulse Responses (RIRs).

9:45

4aAA4. Study of architectural design & acoustic conditions of primary schools in Singapore. Siu Kit Lau and Geok Ling Lee (Dept. of Architecture, National Univ. of Singapore, Block SDE3, #01-06, 4 Architecture Dr., Singapore 117566, Singapore, slau@acousticsresearch.com)

Urban noise problems are a growing concern in rapidly urbanizing cities such as Singapore. Studies have revealed that children are particularly susceptible to the harmful effects of noise. Children in Singapore schools are particularly susceptible considering the long hours spent in naturally ventilated classrooms. This study aims to assess the architectural design and planning of schools in Singapore based on their acoustic qualities. External ambient noise, façade sound insulation, and reverberation time were measured in three schools

about the architectural design to characterize their acoustic environments and seek areas for improvements. It is revealed that all the classrooms studied experience internal noise levels above the guideline values recommended by the World Health Organization (WHO) of 35 dB, suggesting poor acoustic conditions and harmful effects on children's' health, behavior, and learning abilities. Reverberation time standards were only met by the classrooms in one school out of three, implying that speech intelligibility and learning performance are hindered elsewhere. Two overseas case studies are explored to demonstrate how better acoustic environments can be achieved through planning and design in unbuilt schools and abatement strategies for built schools. Architects and planners can adapt and incorporate them into the design of primary schools in Singapore.

10:10–10:25 Break

Contributed Papers

10:25

4aAA5. Active learning classroom configurations and the effects on speech intelligibility. Edwin S. Skorski (Interior Design, Central Michigan Univ., CMU - Human Environ. Studies, EHS Bldg. 228, Mount Pleasant, MI 48859, skors1es@cmich.edu)

For high academic achievement, it is critical that the acoustic environment of classrooms and other learning spaces provide the proper conditions for good speech communication between teacher and students. While there is a significant body of research regarding traditional classroom seating arrangements, less is known regarding the acoustical impacts of "active learning classroom" configurations. Active learning classrooms are typically furnished with mobile tables and chairs that are easily reconfigured in a variety of learning arrangements. Additionally, in an active learning environment, there is typically no fixed position from which the teacher will lecture. While active learning spaces provide flexibility and mobility, which is advantageous for student engagement, they also create a complex acoustic environment where both source and receiver positions have a continually shifting spatial relationship. Using design guidelines found in ANSI/ASA S12.60-2010/Part 1, this study will use computer simulated environments to explore various active learning space configurations and compare the results to generally accepted acoustical performance criteria for RT, C80, STI, and RASTI.

10:40

4aAA6. Results from the summer sound lab: Exploring acoustical options for a multiuse space on campus. Daniel Butko, Zachary Maggia, Collin Abdallah, and Lindsey Trout (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

This paper summarizes the progress of an active multiphase acoustical research project emphasizing implementation of adaptive interactive acoustical responses in multiuse spaces to transform unhealthy learning environments. A multiuse space on campus, which suffers from poor speech intelligibility as a result of excessive reverb and flutter echo revealed through finish materials, room volume, and shape, functions as an architectural sound lab and location for data collection. The project, launched Spring 2017, will span multiple phases and disciplines for data collection, design, prototyping, validation, and construction. Faculty and students are collecting and analyzing acoustical data, adding to a literature review and database of comparable acoustical challenges and results, and validating findings in both computer and physical model form. Collected data will lead researchers to design applications for adaptive and resilient learning environments with minimized acoustical distractions and improved speech intelligibility. Their work will culminate in a final full-scale built form, providing scholarly merit applicable to similar spaces and/or functions. Established industry partnerships will provide development of physical panel-based prototyping, followed by acoustical testing, full-size fabrication, and installation for public interaction and academic validation.

10:55–11:40 Panel Discussion

Session 4aAB

Animal Bioacoustics: General Biosonar

Rolf Müller, Chair
Mechanical Engineering, Virginia Tech, ICTAS Life Sciences District (Mail Code 0917), Blacksburg, VA 24061

Contributed Papers

8:00

4aAB1. Bone conducted sound in a dolphin's mandible: Experimental investigation of elastic waves mediating information on source localization. Michael Reinwald (Lab. for Biomedical Imaging, Université Pierre-et-Marie-Curie, 15 rue de l'Ecole de Médecine, Laboratoire d'Imagerie Biomédicale, Paris 75006, France, michael.reinwald@upmc.fr), Jacques Marchal (Institut d'Alembert - UMR 7190, Université Pierre-et-Marie-Curie, SAINT-CYR-L'ECOLE, France), Lapo Boschi (Laboratoire iSTeP, UMR 7193 UPMC-CNRS, Université Pierre-et-Marie-Curie, Paris, France), and quentin Grimal (Lab. for Biomedical Imaging, Université Pierre-et-Marie-Curie, Paris, France)

Marine mammals, such as dolphins, use audition as a tool to navigate and hunt. It is widely accepted that the main auditory pathway for sound reception in a dolphin's skull is through the lower jawbone and the connected fatty tissues—both possibly acting as a waveguide. Elastic waves traveling inside the mandible therefore likely contain information on the position of sound sources. The objective of the work was to investigate to which extent elastic wave signals in a mandible of a common dolphin (*Delphinus delphis*) can be used to localize a source. Experiments were conducted in a water pool by placing a sound source emitting pulses in the range of 10–100 kHz at different angles in the horizontal plane of the mandible. The elastic waves propagating in the mandible were measured at two positions close to the hypothetical ears by piezoelectric discs glued to the bone. We evaluated energy directivity patterns, interaural level differences, and applied a time-reversal localization algorithm to simulate acoustic source localization of the sound source. The findings should give insight into the directivity of the auditory pathway of sound in a dolphin's mandible and will be substantiated by 3D simulations in future work.

8:15

4aAB2. Temporal features of conditioned reduction in dolphin auditory brainstem responses. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

Conditioned changes in the hearing of two bottlenose dolphins were elicited by pairing a 10-kHz conditioned stimulus (i.e., a warning sound) with a more intense, unconditioned stimulus at 80 kHz. Hearing was assessed by measuring transient auditory brainstem responses (ABRs) to tone burst stimuli presented with inter-stimulus intervals drawn from a random distribution with mean interval of 2 ms and 2-ms jitter. Tone burst frequencies were 57, 80, 113, and 133 kHz. ABRs at discrete times within each trial were obtained by synchronously averaging epochs of the instantaneous electroencephalogram time-locked to tone burst onsets over 2 to 3-s time intervals. In initial testing with one dolphin, ABRs were reduced after the conditioned stimulus but before the unconditioned stimulus, demonstrating a conditioned suppression of the ABR. In subsequent testing, and in all testing with the second dolphin, ABRs were suppressed (relative to baseline values) throughout the entire 31-s trial. When the unconditioned stimulus was removed, ABRs returned to baseline values, demonstrating extinction of the conditioned response. The results support the hypothesis that some toothed whales can "self-mitigate" the effects of noise if warned of an impending exposure. [Work supported by SSC Pacific NISE Program.]

8:30

4aAB3. Range perception in dolphins: Discrimination of sounds with distance-dependent features other than amplitude. Carolyn E. Schlundt (Government IT Services / US Navy Marine Mammal Program, 4045 Hancock St., Ste. 210, San Diego, CA 92110, carolyn.melka@harris.com), Jason Mulsow, Ryan A. Jones (National Marine Mammal Foundation, San Diego, CA), and James J. Finneran (U.S. Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, Code 71510, San Diego, CA)

Although sound pressure level (SPL) is typically used to estimate behavioral responses of marine mammals to anthropogenic sound, other factors, such as source distance, may play a role in an animal's ability to assess the inherent risk of a sound (and thereby affect its response). Factors that contribute to range perception include a decrease in received SPL due to spreading loss and absorption, a relative decrease in the amplitude of high-frequencies from frequency-dependent differences in absorption, and the presence of reverberation due to boundary interactions. In the current study, three bottlenose dolphins (*Tursiops truncatus*) learned to discriminate tones with simulated reverberation and high-frequency absorption (*degraded stimuli*) from tones without those features (*undegraded stimuli*). Tones were frequency modulated, with fundamental frequencies of approximately 10 kHz, and roving SPLs of 120 dB $\pm$ 10 dB. A progression of experiments examined the dolphins' performance when presented with novel, degraded stimuli containing either reverberation or high-frequency attenuation. The results provide experimental evidence that marine mammals can differentially classify sounds with similar SPLs, but differing transmission-related acoustic cues of reverberation and frequency-dependent absorption, and may use signal degradation characteristics to judge the range of acoustic sources. [Work supported by US Fleet Forces Command.]

8:45

4aAB4. Analysis of bats' gaze and flight control based on the estimation of their echolocated points with time-domain acoustic simulation. Taito Banda (Faculty of Life and Medical Sci., Doshisha Univ., 1-3, Miyakotani, Tatara, Kyotanabe, Kyoto 610-0394, Japan, dmq1001@mail4.doshisha.ac.jp), Miwa Sumiya (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe-shi, Japan), Yuya Yamamoto (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), Yasufumi Yamada (Organization for Res. Initiatives and Development, Doshisha Univ., 1-3, Miyako-tani, Tatara, Kyotanabe, Kyoto, Japan), Yoshiki Nagatani (Dept. of Electronics, Kobe City College of Technol., Kobe, Hyogo, Japan), Hiroshi Araki (Adv. Technol. R&D Ctr., Mitsubishi Electric Corp., Amagasaki, Japan), Kohta I. Kobayasi (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), and Shizuko Hiryu (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Bats detect objects by echolocation. With those echoes, they can describe the shape of objects as well as position, and then approach or avoid them. To reveal which part of objects bats gaze at, we simulated the echoes that bats hear, and estimate the bats' echolocated points by time-domain

4a THU. AM

2D-acoustic-simulation based on the behavioral data during obstacle-avoiding flight. First, we constructed a microphone array system in an experimental chamber and attached telemetry-microphones to Japanese horseshoe bats. With that measurement system, we obtained the timing, positions and directions of emitted pulses during obstacle-avoiding flight. Using these data, we simulated the echoes returning from the obstacles (acrylic boards) and investigated how the bats show spatial and temporal changes in the echolocated points of objects as they became familiar with the space. In a comparison between before and after the habituation in same obstacle layout, there are differences in the wideness of echolocated points on objects. By flying same layout repeatedly, their echolocating field become narrower. This study suggests the help of acoustic simulation to understand the way bats see the world. [This research was supported by Scientific Research on Innovative Areas of JSPS, and the JST PRESTO program.]

9:00

4aAB5. Doppler shifts in bat biosonar: The good, the bad, and the unexpected. Rolf Müller, Xiaoyan Yin, and Peiwen Qiu (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu)

Most models for explaining bat biosonar are based on linear systems theory. A notable exception are frequency shifts due to the Doppler effect that occur whenever there is relative motion between the wave emitter and receiver. Bat species such as horseshoe bats, Old World leaf-nosed bats, and mustached bats have evolved particularly sophisticated biosonar systems that are able to exploit Doppler shifts for detecting prey in clutter. In this case, "good Doppler shifts" induced by the wing beat velocity of an insect prey give the prey echoes unique signatures that stand out among echoes from stationary targets such as reflecting facets in dense vegetation. Since these "good Doppler shifts" are small, bats exploiting them had to evolve specialized biosonar mechanisms to detect them. However, these mechanism are also sensitive to "bad Doppler shifts" that result, e.g,. from a bat's own flight velocity. It has been generally assumed that bats with Doppler-based biosonar will eliminate such nuisance Doppler shifts through compensation behaviors to allow them to detect the good Doppler shifts. However, a complete picture of Doppler shifts in bat biosonar systems may also have to account for other sources of Doppler shifts that do not fall into the above categories.

9:15

4aAB6. Integration of dynamic emission and reception in the biosonar system of horseshoe bats. Joseph Sutlive (Translational Biology, Medicine, and Health, Virginia Tech, 3719 Parliament Rd., Apt. 22, Roanoke, VA 24014, josephs7@vt.edu), Hiroshi Riquimaroux (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bats have evolved unique mechanisms by which to navigate and hunt in their environments. Some species of the microchiroptera, such as members of the horseshoe bat family (Rhinolophidae), which live in particularly challenging, cluttered environments, have been observed to move their peripheral structures for emission and reception (noseleaf and pinna) during echolocation very rapidly, e.g., within a tenth of a second. With these behaviors, the bats could create time-variant properties in their biosonar systems to provide additional degrees of freedom to be used for enhancing the encoding of sensory information. To make full use of this opportunity, a tight sensorimotor integration is likely to be required. In order to explore the possible role of sensorimotor integration in a dynamic biosonar system, we have developed biomimetic robotic models to replicate the effects of the dynamics of the baffle structures. Data obtained with these systems indicates that a coordinated emitter and receiver dynamics can both contribute to the embedding of time-variant signatures in the signals that encode information about target class. Ongoing research seeks to transition insights from work on the biomimetic design paradigms to data from brain recordings made in from the auditory brainstem and the superior colliculus of horseshoe bats.

9:30

4aAB7. Quantification of fast pinna motions in rhinolophid and hipposiderid bats. Xiaoyan Yin, Peiwen Qiu, and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, xiaoyan6@vt.edu)

As part of their biosonar behaviors, rhinolophid (family Rhinolophidae) and hipposiderid bats (family Hipposideridae) both show conspicuous motions of their outer ears (pinna). These motions coincide with pulse emission and echo reception in time and could hence have a functional relevance for the encoding of sensory information. However, a quantitative in-depth characterization of these motions is still needed to derive detailed hypotheses for their function. To accomplish this, dense sets of landmark points have been placed on the pinna to provide for dense spatial coverage of its surface over the course of a motion cycle. Occlusion-free stereo tracking of the landmarks was accomplished with an array of four high-speed video cameras. Customized methods based on motion prediction have been used to track landmark points across video frames. The results have been used to construct accurate, continuous estimates of pinna surface motion. These estimates reveal that the pinna surface is subject to heterogeneous patterns of displacements and velocities within each motion cycle. Experiments with simplified robotic reproductions to understand the acoustic implications of these pinna surface motions are currently in progress. Once the signal transformations that result from the pinna motions are understood, the question of functional relevance can be addressed.

9:45

4aAB8. Noseleaf motions impart dynamic signatures on bat biosonar pulses. Liujun Zhang, Ru Zhang (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Shanda South Rd. 27, Jinan 250100, China, sdujune@gmail.com), Shuxin Zhang (Shandong University - Virginia Tech Int. Lab., Shandong Univ., Jinan, Shandong, China), Luhui Yang, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Old-World leaf-nosed bats (Hipposideridae) are echolocating bats with nasal biosonar pulse emission. The nostrils in these animals are surrounded by baffle shapes ("noseleaves") that have been shown to function as beamforming devices. In addition to their elaborate static geometries, the noseleaves can also undergo non-rigid shape changes that coincide with biosonar pulse emission. Prior work by the authors based on biomimetic baffle shapes has demonstrated that baffle motions similar to the ones seen in bat noseleaves can result in time-variant device properties, i.e., distributions of signal power over direction and frequency that also change with time. However, it remained to be established whether similar effects occur in bats. To answer this question, biosonar pulse emission in hipposiderid bats (*Hipposideros armiger* and *Hipposideros pratti*) has been studied with synchronized arrays of high-speed video cameras and ultrasonic microphones. Noseleaf motion was characterized by tracking the position of nine landmark points on the noseleaf. Temporal variability in the distribution of signal energy across spatial directions and ultrasonic frequencies was quantified using pair-wise measures of similarity applied to time-windowed segments of the pulse waveforms. Pulses that were accompanied by unequivocal noseleaf motions showed a significantly larger variability across time than those that were not.

Session 4aAO

Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics: Biological Effects on Seabed Geoacoustic Properties I

Kevin M. Lee, Cochair
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Kelly M. Dorgan, Cochair
Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528

Chair's Introduction—8:00

Invited Papers

8:05

4aAO1. Some startling effects of biologic factors on seafloor physical properties. Thomas F. Wever (WTD71, Eckernförde, Schleswig-Holstein, Germany) and Chris J. jenkins (INSTAAR, Univ. of Colorado, Boulder, 4001 discovery drv, Boulder, CO 80309-0450, jenkinsc0@gmail.com)

Recently, the biological impact on marine geophysics has received much interest. The question now is What types of impacts, how intense, and how changing? How can these impacts be understood and quantified, and then enacted in our practical predictive systems? Two dramatic examples from NW Europe. The acoustic response of the sandy seabeds is being altered by dense populations of species such as the invasive Ensis (i.e., a change of seabed properties). The same species forms a baffle that alters the seafloor response to flows, causing sediment trapping (i.e., a change of processes). At the sea surface, massive drifting rafts of detached bladder algae change the acoustic response of the sea surface and can attenuate surface waves. The rafts have seasonal and longevity attributes and can be tracked remotely, so effects on long-range underwater communications may be manageable. The Volkswagenstiftung, Office of Naval Research, NATO, and others have understood the significance and started to finance meetings, projects, and expert teams. To improve prediction capabilities, especially before at-sea operations, a close cooperation of biology and physical sciences is necessary. This can be achieved by two parallel approaches: (1) make existing bio- and geo-information compatible and (2) design new combined bio-geo experiments.

8:25

4aAO2. The trait-based approach as a powerful tool to study the bio-geo seafloor system. Anna Törnroos (Environ. and Marine Biology, Åbo Akademi Univ., BioCity, Tykistönkatu 6, Turku 20520, Finland, anna.m.tornroos@abo.fi)

One of the major causes of heterogeneity on the seabed is the biology. The presence of organisms creates voids and frameworks within and on the sediment, and their behavior may layer or sort the entire seafloor. Making use of the biological information would be powerful for improving acoustics. Likewise, embracing, to a greater degree, acoustic techniques and measurements to understand the biology would be favorable. However, to tackle this bio-geo diversity in a cross-disciplinary way requires a common language and approach. Here, I present and propose the trait-based approach as a way forward. Because there are simply too many species to describe and include in one model, reducing this complexity is essential and can be done by considering individuals characterized by a few key characteristics, or *traits*. Relevant biological traits span morphology, behavior and life-history of organisms and can be applied on single individuals and scaled up to whole communities, incorporating the density of organisms. By exemplifying the progress in benthic ecology, I outline where we currently stand, possible key traits of value for both fields and ideas on how to progress.

8:45

4aAO3. Relating functional diversity of infauna and sediment geoacoustic properties. W. Cyrus Clemo and Kelly M. Dorgan (Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, wcc1622@jagmail.southalabama.edu)

Marine sediments cover most of the seafloor and vary in physical structure from unconsolidated gravel or sand grains to cohesive, gel-like muds. Sediments also provide a habitat to diverse and abundant animals (infauna). Infauna are impacted by but can also affect the sediment's physical and biogeochemical properties and fluid flow through and across the sediment. Through activities such as burrowing, feeding, and construction of tubes and rigid body parts, many infauna act as ecosystem engineers, directly and indirectly

creating and changing their habitats. Characterization of infauna using functional groups based on how their activities affect their sediment environments can simplify the broad diversity of infauna. For example, some infauna create habitat structure in addition to modifying surface topography through tube construction. Others change sediment properties through bioturbation (animal-induced particle mixing) and bioirrigation (fluid mixing) as they burrow. Structure creation and bioturbation/irrigation can affect sediment geotechnical properties such as porosity, bulk density, grain size heterogeneity, fracture toughness, and stiffness. Changes in these properties can influence acoustic wave speed, attenuation, and backscatter. Thus, acoustics tools can be used to observe and quantify animal-sediment interactions relatively noninvasively by relating the activities of different functional groups with changes in acoustic properties of sediments.

9:05

4aAO4. Biogenic control on rheological and geoacoustic properties of aquatic sediments: Past research and future directions. Samuel J. Bentley (Geology and Geophys. and Coastal Studies Inst., Louisiana State Univ., E235 Howe Russell GeoSci., Dept. of Geology, LSU, Baton Rouge, LA 70803, sjb@lsu.edu)

Our understanding of biogenic controls on rheological and geoacoustic properties of aquatic sediment lags behind other topics in seafloor science. For nearly a century, the ability of sedimentary infauna to influence fabric and diagenesis of aquatic sediments has been widely recognized. These insights result from many studies (both observation and modeling) of bioturbation and its control on mass transport and destruction of primary sedimentary layering in the shallow seabed. Nevertheless, our detailed knowledge infaunal behavior below the intertidal zone remains very limited for most taxa. Our understanding of infaunal controls on sediment physical properties and rheology such as shear strength, porosity, or sound-wave propagation below the intertidal zone remains even more sparse. Some pioneering studies over the decades have demonstrated that biogenic influences on sediment porosity, strength, and rigidity can be significant and are related to bioturbation style and rate. However, the number of these studies remains small compared to the breadth of the problem. Several recent numerical models of muddy-sediment seabed dynamics and consolidation hold promise as frameworks for potential experimentation, in which biogenic forcing can be implemented to dilate, compact, or reshape sediment, and so alter sediment rheological and geoacoustic properties. This presentation will explore potential new approaches in this line of research, based on past successes and present knowledge gaps.

9:25

4aAO5. Fast-marching methods to model the acoustic responses of biological seafloors. chris j. jenkins (INSTAAR, Univ. of Colorado, Boulder, 4001 Discovery Dr., Boulder, CO 80309-0450, jenkinsc0@gmail.com)

Biologically colonized seafloors have been difficult to model for their acoustic interaction. The distributions of geoacoustic properties above and beneath the nominal bottom can be extremely complex. We have implemented workflows in software which: (i) efficiently construct many patterns of growth forms, bioturbation, organism aggregations, and flesh-skeleton morphologies, and (ii) submit them to "Eikonal" wavefront acoustic modeling. Ray-based finite-element and statistical methods can, in theory, be applied but they run into problems of the strongly inhomogeneous geoacoustics, ray-chaos (of several causes), absence of a clear division into the usual rough surface/inhomogeneous volume halfspaces, and heavy computational costs. Wavefront methods, namely, Eikonals implemented via Fast-Marching Algorithms, are able to deal with complex situations—as has been realized lately in deep seismic geophysics, in moving object tracking, and interestingly, in urban settings, for locating gunshots and mapping Wi-fi fields. They also to offer a new way to understand the acoustic backscatter results from multibeam, sidescan and single beam sonar systems over complex biologic seafloor types. This direction of research requires close knowledge of both the "bio" and the "geo" of seafloors.

9:45

4aAO6. Do environmental stress-induced changes in faunal composition and bioturbation affect sediment geoacoustic properties? A test case in the Gulf of Mexico dead zone. Kevin Briggs (Div. of Marine Sci., Univ. of Southern Mississippi, 1020 Balch Blvd., Stennis Space Ctr., MS 39529, Kevin.B.Briggs@usm.edu), Michael Richardson (Inst. for Defense Analyses, Alexandria, VA), and Shivakumar Shivarudrappa (Louisiana Universities Marine Consortium, Chauvin, LA)

Four sites on the continental shelf off Louisiana, each subjected to different historical exposures to low concentrations of bottom-water dissolved oxygen, are investigated in terms of their macrobenthos species composition, sediment physical properties, and sediment geoacoustic properties (sound speed, attenuation, and impedance). From macrobenthos species composition, feeding type is identified, which allows categorization of some bioturbation activity as either dilation or compaction of sediment. Dilation and compaction should affect sediment properties of bulk density and porosity, which are significant predictor variables of geoacoustic properties. Different levels of oxygen stress correspond with statistically separable macrobenthos assemblages, abundance and diversity of biogenic structures (burrows and voids), and ratios of dilators to compactors. Sediment sound speed and attenuation values measured in subcores from box cores are compared at each of the four sites with different faunal composition and average bottom-water dissolved oxygen concentrations to test the long-term (historical) effects of different bioturbation regimes on geoacoustic properties. Sediment acoustic properties are regressed on sediment physical properties and compared with established empirical fits from a worldwide database to identify possible anomalies. The impedance of surficial sediment is measured at each site with the Acoustic Sediment Classification Profiler to examine possible large-scale (side-wide) effects on geoacoustic properties.

10:05–10:25 Break

10:25

4aAO7. Time-dependent seafloor roughness simulations using a coupled spectral ripple-biological decay model. Allison Penko and Joseph Calantoni (Marine GeoSci. Div., U.S. Naval Res. Lab., 1005 Balch Blvd., Code 7434, Stennis Space Ctr., MS 39529, allison.penko@nrlssc.navy.mil)

Ever changing hydrodynamic conditions in the nearshore continuously affect biological communities and sediment features on the seafloor. Seafloor sediments have complicated interactions with the co-existing biological flora and fauna on the seabed. For example, ripple formation by waves and ripple degradation from biological activity are opposing forces occurring continuously. Simulating this complex natural environment is challenging due to these dynamic interactions of processes that span across a wide range of length and time scales. Existing seafloor models are often based on equilibrium conditions and neglect any change due to biological effects. Additionally, typical equilibrium ripple models provide only geometric dimensions, not a roughness spectrum necessary for acoustic applications. We present a time-varying spectral seafloor model, NSEA, that predicts the spatial and temporal evolution of seafloor roughness accounting for both the evolution and degradation of ripples due to hydrodynamics and biology. Model predictions are compared to high-frequency sector-scanning sonar data collected off the coast of Panama City, FL, during the Target and Reverberation Experiment (TREX13) and at the US Army Corps of Engineers Field Research Facility in Duck, NC. The sector-scanning sonar imaged ~15 square meters of the seafloor every 12 min for up to four weeks in depths up to 20 m.

10:45

4aAO8. Predicting acoustic backscattering from bioturbated seafloors. Shawn Johnson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, shawn.johnson@psu.edu)

Seabed roughness is an important property for high-frequency scattering. However, for certain littoral environments, the surficial roughness of the seabed is often continuously being modified by hydrodynamic and biologic forces. For example, ocean surface waves may create orbital ripples on sand seafloors which are characterized by anisotropic roughness, and acoustic scattering from such a seafloor would be strongly aspect-dependent. Yet bottom-dwelling organisms rework the sediment, returning the structured seabed relief to random isotropic roughness, where acoustic scattering is more uniform with respect to the aspect ensonified. In this talk, we will utilize both analytic expressions as well as procedural generation methods to predict the two-dimensional roughness equilibrium power spectra of rippled and bioturbated seafloors, and the resulting impact on acoustic scattering prediction.

11:05

4aAO9. Sonar observations of biological activity in marine sediments. Darrell Jackson (Appl. Phys., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

This talk will begin with a review of past experiments using bottom-mounted, rotating sonars to observe temporal change in the seafloor. These measurements were conducted at 40 kHz and 300 kHz, and provide images of change as represented by ping-to-ping correlation. Time scales of change vary greatly with frequency and sediment type, with changes occurring over weeks at 40 kHz at a silty site and over minutes at 300 kHz at a sandy site. Finally, the level of scattering due to relic tubes, shells, etc., will be compared to that of live animals. The conclusion is that it is easiest to observe the cumulative effects of biological activity, but techniques are available to isolate current activity.

11:25

4aAO10. Local suppression of bioturbation and its acoustic consequences. Brian T. Hefner, Steven G. Kargl, and Kevin Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

Seafloor roughness in shallow waters can be both spatially and temporally dynamic. While wind-driven waves and bottom currents can generate bedforms such as ripples on sand sediments, bottom-feeding fish tend to limit the lifetime of these anisotropic features. This greatly reduces the time during which high frequency sonars can exploit the ripples to coherently penetrate the sediment at subcritical angles. Sonar performance prediction in these environments therefore relies heavily on sediment transport models to generate roughness predictions that accurately reflect changes on these time scales. The competition between the hydrodynamics and biology of the environment can be further complicated by the natural introduction of materials which can locally suppress the bioturbation. Two examples are discussed here: The first comes from the Sediment Acoustics Experiment in 2004 where a storm deposited a layer of mud over some areas of seafloor protecting the bedforms from fish. The second is from the recent Target and Reverberation Experiment where there is evidence that spatially-varying shell content can also reduce the effects of bioturbation. The consequences for sonar performance in these two cases are discussed. [Work supported by the U.S. Office of Naval Research and the Strategic Environmental Research and Development Program.]

Session 4aBA

Biomedical Acoustics: Ultrasound-Mediated Neuromodulation

Parag V. Chitnis, Chair
Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Chair's Introduction—8:25

Invited Papers

8:30

4aBA1. Ultrasound neuro-stimulation effects of peripheral axons *in-vitro*. Nader Saffari, Christopher J. Wright (Mech. Eng., Univ. College London, UCL, Torrington Pl., London WC1E 7JE, United Kingdom, n.saffari@ucl.ac.uk), and John Rothwell (UCL Inst. of Neurology, Univ. College London, London, United Kingdom)

Neurological disorders are a huge disease burden for individuals and the economy worldwide. For a few of these disorders, successful new treatments have been found in neuro-stimulatory techniques. Ultrasound (US) can target the brain in a spatially precise, non-invasive manner, stimulating or suppressing neuronal activity. Much success has been reported with US in small animals, primates, and human subjects. Despite all the recent progress, very little is known about the molecular and cellular mechanisms behind the observed neural responses. This study uses a controlled *in-vitro* environment, directly stimulating and recording CAPs from excised crab nerves (*Cancer pagurus*). The aim is to create an environment where both the biological and ultrasound environment can be measured and modelled accurately to gain insight into the mechanism by which mechanical forces are transduced into propagating electrical activity in nerve fibres. The results demonstrate that the constituents of unmyelinated axonal tissue are sufficient to generate de-novo action potentials in response to US stimulus. The threshold for this stimulation was much higher than similar procedures performed on CNS models but in good agreement with other PNS focused studies. They also provide the first clear evidence for the involvement of cavitation as an ultrasound stimulation mechanism.

9:00

4aBA2. Ultrasonic stimulation and metabolic stress in neuronal systems. John R. Cressman (Phys. and Astronomy, George Mason Univ., 4400 University Dr., MSN2a1, Krasnow Inst., Fairfax, VA 22030, jcressma@gmu.edu), Monica Gertz (Neuroscience, George Mason Univ., Fairfax, VA), and Parag V. Chitnis (Dept. of BioEng., George Mason Univ., Fairfax, VA)

We report on computational and experimental investigations into the effects of high-frequency focused ultrasound stimulation on neuronal tissue. We simultaneously measuring local field potentials as well as extracellular oxygen and potassium concentrations in hippocampal brain slices to investigate physiological responses, including metabolic demand, during ultrasonic stimulation. To better interpret our observations, we utilized computational models that incorporate dynamics for transmembrane ionic and osmotic flows, membrane fluctuations, and metabolism. Our experiments and models suggest that ultrasonic stimulation can produce substantial ionic redistribution that leads to a significant metabolic burden.

Contributed Paper

9:30

4aBA3. Imaging of tissue displacement induced during focused ultrasound neuromodulation *in vivo*. Stephen A. Lee, Matthew Downs, Niloufar Saharkhiz, Yang Han, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 500 W 120th St., New York City, NY 10027, sal2212@columbia.edu)

Our group has recently shown feasibility of FUS modulation of the peripheral nervous system (PNS) *in vivo*. However, the interplay between FUS and the PNS is not completely understood and has never been imaged. To unveil both the mechanism as well as provide an image-guided approach to modulation monitoring, a new modulation system was designed to simultaneously image the mechanical perturbation of the tissue during modulation *in vivo*. The system consisted of a 4.5 MHz HIFU confocally aligned with a 7.8 MHz imaging transducer. Activation of the sciatic nerve of the mouse was induced with parameters previously reported for successful modulation. 200 RF frames at a 10 kHz pulse repetition frequency were used for 1D cross-correlation (20 lambda window, 90% overlap) to calculate the inter-frame axial displacement before, during, and after modulation. Displacement maps overlaid on the B-mode images illustrate that once FUS is applied, downward displacement was detected where the highest displacement is located at the focus (9.8 micron average peak displacement). After FUS modulation, displacement steadily reduced to baseline (0.5–0.8 ms). Our findings indicate that FUS neuromodulation is associated with the radiation force effect and therefore successful application is dependent upon sufficient displacements induced.

9:45–10:00 Break

Invited Papers

10:00

4aBA4. Intersections of neuromodulation, focused ultrasound, and gene delivery with brain-penetrating nanoparticles. Brian Mead (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908), Namho Kim, Karina Negron (Johns Hopkins Univ., Baltimore, MD), Wilson Miller (Radiology, Univ. of Virginia, Charlottesville, VA), Panos Mastorakos, Jung Soo Suk (Johns Hopkins Univ., Baltimore, MD), Alexander Kilbanov (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), Justin Hanes (Johns Hopkins Univ., Baltimore, MD), and Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA, rprice@virginia.edu)

Focused ultrasound (FUS) can modulate CNS tissue function by facilitating MR image-guided transfection of selected brain structures with therapeutic genes. Conversely, we hypothesize that modulation of CNS tissue with FUS may be used to enhance transfection. In previous studies, our group has (i) transfected selected brain structures by delivering systemically administered, non-viral, "brain-penetrating" nanoparticles (BPN) across the blood-brain barrier (BBB) with FUS and microbubbles (MBs) and (ii) employed this platform to restore dopaminergic motor neuron structure and function in a rat Parkinson's disease model via delivery of a neurotrophic factor (GDNF) gene to striatum. Here, we show that pre-conditioning of rat neural tissue with pulsed FUS or pulsed FUS + MBs before intrastriatal convection-enhanced delivery of BPN complexed with a reporter gene (CMV-ZsGreen) serves to enhance transfection volume by 36% or 151%, respectively. Pre-conditioning with both FUS and FUS + MBs proved effective in mice as well, with both FUS protocols yielding ~75% increases in transfection. We conclude that modulation of CNS tissue plays an important role in dispersing BPN after they are delivered across the BBB using pulsed FUS and MBs. Pre-conditioning of CNS tissue with FUS may eventually represent an attractive means for enhancing transfection.

10:30

4aBA5. Central and peripheral nervous system modulation using ultrasound. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Focused ultrasound (FUS) neuromodulation has previously been proposed as a promising technique to drive neuronal activity and has been shown throughout a breadth of applications including in mice, rats, non-human primates, and humans as a novel technique for the noninvasive manipulation of neuronal activity using ultrasound. Our group has demonstrated excitation of both the central (CNS) and peripheral nervous system (PNS). In the CNS, motor- and cognitive-related brain regions of mice were induced by targeting specific brain structures. Higher acoustic pressures increased the success rate. Pupil dilation was observed when neuromodulating regions in the brain covering the superior colliculus and other anxiety-related structures such as hippocampus and locus coeruleus. In the PNS, we showed for the first time stimulation of the sciatic nerve with FUS eliciting a physiological motor response was recorded *in vivo*. Clipping the sciatic nerve downstream of stimulation eliminated EMG activity during FUS stimulation. Peak-to-peak EMG responses and delay to signal were comparable to conventional electrical stimulation methods. Histology along with behavioral and thermal testing did not indicate damage to the nerve or surrounding regions. Our studies demonstrated the capability of FUS to modulate target specific regions in both the brain and the peripheral *in vivo*.

Contributed Paper

11:00

4aBA6. Ultrasonic neuromodulation of the mouse cortex increases cardiorespiratory activity. Christian Aurup and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., P&S 19-418, New York, NY 10032, ca2625@columbia.edu)

Focused ultrasound (FUS) has demonstrated its ability to modulate neuronal activity in both cortical and subcortical brain regions in a noninvasive and safe fashion in mice, non-human primates, and humans. Rodent studies have shown that ultrasonic neuromodulation (UNMOD) can elicit motor responses in limbs and trigger pupil dilation. Little is known about the effect of ultrasonic neuromodulation of the CNS on the autonomic activity in mice, primarily due to the cardiorespiratory depression caused by the anesthesia used. However, urethane has limited effects on autonomic activity and brain hemodynamics. In this paper, we demonstrate that UNMOD of the cortex increases cardiorespiratory activity in mice. A 1.9-MHz single-element focused ultrasound transducer was used to apply pulsed ultrasound to the mouse cortex. Each animal was injected intraperitoneally with urethane (1500 mg/kg), allowing for a stable plane of anesthesia and experimental efficacy window often exceeding 4 hours with minimal effects on autonomic activity. Sonications were performed in a grid spanning the cortex centered at the Bregma skull suture and heart and breathing rates were acquired using a pulse oximeter. FUS resulted in significant increases in both breathing and heart rates immediately following sonication. This study demonstrated that FUS can have an autonomous excitatory effect *in vivo*.

Invited Paper

11:15

4aBA7. High resolution modulation of human brain circuits using transcranial focused ultrasound. Wynn Legon (Neurosurgery, Univ. of Virginia, 426 Church St., SE MMC 8388, Minneapolis, MN 55406, wlegon@umn.edu)

Current non-invasive electric and electromagnetic non-invasive neuromodulatory approaches have proven effective for inducing transient plastic changes in human cortex. However, these technologies have poor spatial resolution and suffer from a depth-focality tradeoff. Transcranial focused ultrasound (tFUS) is an emerging non-surgical low-energy technique for inducing transient plasticity in sub-cortical and cortical areas with high spatial resolution and adjustable focus. tFUS has been successfully employed in small and large animal preparations and our work has also demonstrated tFUS to be effective for human cortical and subcortical neuromodulation. Here, we will discuss tFUS current approaches as well as challenges and future directions for this technology as a tool for neuroscience research and human brain mapping and potential clinical applications.

THURSDAY MORNING, 7 DECEMBER 2017 STUDIO 2, 7:55 A.M. TO 12:00 NOON

Session 4aNS

Noise and ASA Committee on Standards: Urban Planning Using Soundscape II

David Woolworth, Cochair
Roland, Woolworth & Associates, 365 CR 102, Oxford, MS

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

Chair's Introduction—7:55

Invited Papers

8:00

4aNS1. Urban design with noise in mind—A model for future ASA outreach workshops for city planners and decision makers. Kerrie G. Standlee (DSA Acoust. Engineers, Inc., 15399 SW Burgundy St., Tigard, OR 97224, stanhartk@comcast.net)

In early 2009, the Acoustical Society of America funded the development of materials that could be used to introduce urban planners and decision makers to the concept of using soundscaping as a tool in land-use planning. After several months of discussion among interested persons, a symposium outline was generated and speakers were solicited to develop materials that would address specific topics considered important in training urban planners. The materials were developed and first used at a symposium co-sponsored by the ASA and the City of Portland during the ASA May, 2009 meeting in Portland, Oregon. The materials were again presented at a symposium co-sponsored by the ASA and the City of Baltimore during the ASA April, 2010 meeting in Baltimore, Maryland and finally at a symposium co-sponsored by the ASA and the City of Seattle during the ASA May, 2011 meeting in Seattle, Washington. This paper presents a summary of the materials used at the three symposiums and discusses how the material can continue to be used as an ASA outreach event at future meetings around the country.

8:20

4aNS2. An exploration of the urban design possibilities offered by soundscape theory. Gary W. Siebein, Hyun Paek, Gary Siebein, Keely Siebein, Marylin Roa, and Jennifer R. Miller (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

The basic concepts of soundscape theory including soundmarks which are the acoustical equivalent of landmarks, keynote sounds which are the typical sounds in a vicinity; and sound signals or the specific acoustic events that comprise the ambient sound were presented by Schafer (1977) and Truax (2001). These concepts offer ways to document, analyze, and design the acoustical qualities of urban places similar to the way that the concepts developed by Kevin Lynch in the Image of the City (1960) did for the visual character of cities. The urban design potential of soundscape theory lies in identifying the ecological relationships linking sources of sounds,

prospective listeners of sounds, and the "coloration" of the sounds by the urban environment that gives unique identities to localities within a city. An acoustical community is formed by the parties who are related by the need for various forms of tangible and intangible communication in an urban setting. Case studies of an entertainment district in a city, the initial planning of a new town and the redevelopment of a city park will be used to illustrate the methods and applications of this theory to enhance the aural and urban design of cities and towns.

8:40

4aNS3. Quietness as a commons: Integrating soundscape in urban planning for the environmentally just city. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de)

Today, urbanization and environmental pollution represent major issues of the urban planning agenda and one of the most relevant challenges is constituted by planning environmentally and socially just cities to protect human health and ensure quality of life and well-being. To achieve this goal, noise pollution and the scarcity of quietness in cities have also to be taken into account. In Europe, the importance of quiet areas has been first recognized by the 2002 Environmental Noise Directive and, since then, numerous projects have been developed by the EU Member States to define a common methodology to protect quiet areas. Notwithstanding, according to the European Environment Agency, there is still the need for in-depth research in the field. This paper argues that integrating soundscape in urban planning processes, through the paradigm of "quietness as a commons," could significantly contribute to fill this gap of knowledge. This assumption is discussed by presenting a novel citizen-driven methodology to analyze, assess and plan urban quiet areas, implemented in a pilot study in Berlin. In detail, this paper illustrates the methods applied, the findings and the first draft of planning guidelines developed with the community to protect existing quiet areas in the pilot area.

9:00

4aNS4. Basic concepts on sound sculptures design in soundscapes. Fernando J. Elizondo - Garza (Acoust. Lab., FIME-UANL, Calle 8 #422, Col. Villazul, San Nicolas, N.L. 66420, Mexico, fjelizon@hotmail.com) and Cesar Guerra -Torres (Acoust. Lab., FIME-UANL, Monterrey, N.L., Mexico)

The primitive development of visual sculptures was born for the rite and evolved toward the artistic object, that in parallel with the development of music and musical instruments. Currently with a broad sense conceptualization of the art object, the sculpture has evolved from an aesthetic object towards the concept of installation, a multisensory interactive act, within which the sound sculpture, or installation, has taken predominance. In this paper some of the basic (mental) conceptual schemes to be considered in the design of the acoustic dimension of sound sculptures will be presented, e.g., the sculpture as a sound source and its acoustic field; its location and their primary and secondary effects; acoustic or audio; the to think-design-build process in that order, among others. Some examples of the various types of objects that can be considered sound sculptures will be shown and the importance of the sound sculptures in the design of soundscapes will be discussed.

9:20

4aNS5. Soundscapes in historic urban environments—A case study. Pamela Jordan (HEAD-Genuit-Foundation, Technische Universität Berlin, Institut für Strömungsmechanik und Technische Akustik Fachgebiete der Technischen Akustik, Sekr. TA7, Einsteinufer 25, Berlin 10587, Germany, pam.f.jordan@gmail.com)

In current soundscape practice, soundscape descriptors tend to focus on present conditions, either toward an assessment of site conditions or as predictive tools for future interventions. But does the same approach work when the acoustic environment reflects the past as well, such as at historic sites? How does the complexity of a place representing both past and present inform visitors' reactions to the soundscape? Results from a recent study at the Berlin Wall Memorial will be presented, where a guided soundwalk, informed by archival accounts of the area, was combined with a survey to scrutinize visitor impressions of the soundscape in conjunction with the site's sonic past. Descriptors common to soundscape studies were used in semantic scales alongside new proposals aimed directly at understanding the role that historic information plays in assessments by visitors and local experts. Results from the study will be presented, showing how historic information can subvert some of our fundamental assumptions about soundscape, such as what constitutes "appropriate" or "pleasant" conditions, and how results may differ across expert groups. Indeed, this research demonstrates the powerful impact that historic context can exert on soundscape assessment, with implications for soundscape practice, heritage interpretation, and urban planning alike.

9:40

4aNS6. The advantage of binaural recording for soundscape assessment. Klaus Genuit (Head Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

In the beginning, binaural recording was used for music productions. For more than 40 years, applications are being intensively used in the automotive industry to optimize vehicle interior sound. Only a binaural recording system like an artificial head enables listeners using calibrated equalized headphones to perceive hearing events which are very realistic to the original sound field. Nowadays, within the scope of soundscape standardization ISO 12913, recommendations are being discussed regarding the use of binaural technology for recording environmental sounds, where in the past normally only single microphones were used. For an overall estimation of the total sound pressure level, this may be sufficient, but in a complex sound situation, like soundscapes with several different sound sources at different locations, the selectivity of the human hearing is only given properly by binaural listening. This paper will explain the technology of binaural recording and playback systems and its advantages in comparison with other sound recording systems.

10:00

4aNS7. Evaluation of urban soundscapes: Field vs. laboratory assessments using binaural recordings. Roland Sottek and André Fiebig (Head Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

It is well-acknowledged that soundscape investigations must be carried out in the original context. In order to investigate the influence of acoustic cues on the perceptual measurements, binaural recordings were performed during soundwalks in the center of Gothenburg (Sweden) using a calibrated binaural headset. The participants were 20 students from the Chalmers Technical University. The evaluations were carried out *in-situ* regarding, e.g., loudness, appropriateness, pleasantness, and eventfulness at eight different outdoor gathering spaces. The group of students was divided into four groups. Two groups went from location 1 to 8, separated in time by 15 minutes. The other two groups went from location 8 to 1. At all eight locations each group recorded the acoustic environment (overall 32 recordings). The laboratory assessments were performed with the same students the next day using four recordings measured at four different locations. First, the evaluations were performed simultaneously in the classroom using Sennheiser HD 414 headphones, and second in groups of four students using four calibrated binaural headsets in playback mode. The paper describes the experiments performed in the original and laboratory context, and discusses the results of the different approaches. Additionally, instrumental parameters such as representative values for loudness are compared to the perceptual results: a relevant aspect regarding the second part of the international standard on soundscape dealing with data collection.

10:20–10:40 Break

10:40

4aNS8. e-Appraisal of soundscape for public squares in China. Andy Chung (Smart City Maker, Smart City Maker, Copenhagen, Denmark, ac@smartcitymaker.com) and W. M. To (Macao Polytechnic Inst., Macao, Macao)

There have been activities undergoing in public squares in China changing the soundscape of the venues as well as their neighborhood. Such activities include group dancing with music, martial arts playing sound generating tools, etc., which are not the anticipated activities during the planning and design stage. Soundwalk and surveys using an app were conducted to have an appraisal of the situation. This paper presents the findings of the case studies and recommendations for future planning of public squares with unanticipated sound generating activities.

11:00

4aNS9. Going beyond noise in urban planning—Human perception will be the trusted guide. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de) and Andrè Fiebig (Head Acoust., Herzogenrath, Germany)

When it comes to urban planning "people's perceptions or experiences and/or understanding of an acoustic environment" should be the guiding feature. The use of a variety of data collection methods related to human perception, acoustic environment and context is recommended according to ISO 12913. Following this standard, it is human perception which is paramount determining and stimulating any physical measurement. The international standard ISO 12913 will allow the collection of perceptual data in a harmonized way to get access to the key components in Soundscape: people, acoustic environment, and context. This paper will introduce and discuss techniques of interviews and guidelines, exploration of areas through soundwalks, but also the collaboration platform within a soundscape approach regarding urban planning. It will present strategies for "measurement by persons," and those applied in "measurement by instruments" to overcome the significant gaps that still exists between the both.

11:20

4aNS10. Sensitivity to sounds produced in the city, sound ordinance issues, and cultural planning in New Orleans. Helene Stryckman (Universite Catholique de Louvain, Ave. du Haut-Pont 18, Brussels 1050, Belgium, str.helene@gmail.com) and David Woolworth (Roland, Woolworth & Assoc., Oxford, MS)

In an effort to address sound ordinance issues and better quantify and qualify the soundscape of New Orleans, multiple soundwalks have been held in the older areas of the city over the last few years, specifically in the densest tourist areas (Vieux Carre/French Quarter and Marigny). Sound level data and participant response from several soundwalks are considered with respect to age, sex, occupation, stakeholder status, and language of descriptive used by the respondents. This soundscape inquiry technique has been complemented by qualitative ethnographic fieldwork performed in 2015 that describes the sound ordinance issues and the communities historically involved: musicians, music venues owners, cultural advocates, newcomers, and neighborhood associations. This qualitative case study reveals a process-based dynamic that evolves around different stakeholders' sensitivity to sounds produced in the city. We propose that a sound ordinance should attempt to be neutral in regard to cultural effects and be based on the community-asset knowledge of local experts.

11:40

4aNS11. The soundwalk as a tool to improve community relations and aid in (sound) conflict resolution. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Helene Stryckman (Universite Catholique de Louvain, Brussels, Belgium)

The measurement of soundscapes (under development as ISO/DIS 12913-2) is often performed with local experts in the form of a soundwalk, and for the sake of comparable results requires standardization. One issue stemming from the standardization of the soundwalk is its applicability and usefulness to all situations/locations from a holistic point of view (social, cultural, political, etc.) and whether it considers all significant factors that affect the participants. This paper examines potential benefits of a less structured approach that permits/encourages interaction between the participants who may be at odds in regard to their views on the soundscape, and also provides observations on the impressions of repeat participants. Additionally, several observations of non-auditory or personal factors from soundwalks will be provided. The work referenced in this paper was performed in New Orleans.

Session 4aPA

Physical Acoustics: Infrasound, Atmospheric Sound Propagation, and Turbulence

Roger M. Waxler, Chair
NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Contributed Papers

9:00

4aPA1. Infrasound from small explosions from near to intermediate ranges. Luis De Jesus Diaz, Richard D. Costley, Sarah McComas, Christopher Simpson, James Johnson (Geotechnical and Structures Lab., U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, Luis.A.DeJesus-Diaz@usace.army.mil), and Chris Hayward (Dept. of Geophys., Southern Methodist Univ., Dallas, TX)

An experiment was performed near the U.S. Army Engineer Research and Development Center (ERDC) site near Vicksburg MS on May 2014. Explosive charges were detonated and the shock and acoustic waves were detected with pressure gauges, infrasound sensors, and seismometers stationed at various distances from the source, from 3 m to 15 km. One objective of the experiment was to compare the effectiveness of different wind filter strategies. Toward this end, several sensors were deployed near each other, approximately 8 km from the site of the explosion. These sensors used different types of wind filters, including different length of porous hoses, a bag of rocks, a foam pillow, and no filter. Signal-to-noise estimates made from signals recorded with these different sensors will be used to evaluate the effectiveness of the different strategies. A second objective was to compare the infrasound and seismic signals recorded with collocated infrasound sensors and geophones. Results from this experiment will be presented. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.

9:15

4aPA2. Effects of simplified atmospheric profiles on short range infrasound propagation. Andrew Lammers (Atmospheric Sci., Univ. of Illinois at Urbana-Champaign, 1301 W Green St., Urbana, IL 61801, alammers719@gmail.com) and Michelle E. Swearingen (US Army ERDC, Champaign, IL)

Meteorological profiles strongly influence the propagation of infrasound signals. Much of the infrasound literature has focused on long-range (>250 km) propagation; however, there is interest in short-range (<150 km) propagation from an infrastructure monitoring standpoint. Therefore, understanding the effects of meteorological profiles is crucial. This paper focuses on simplified vertical temperature and wind profiles up to 20 km altitude and their effect on an infrasonic signal emanating from an arbitrary point source. A wide-angle, finite-element PE model that correctly handles discontinuities in wavenumber is used for calculating transmission loss. A large number of simulations were performed to investigate the effect that temperature and wind profiles in different layers of the atmosphere have on surface transmission loss and to assess the sensitivity of varying these profiles. A discussion of the results of these simulations are presented as well as an overview of temperature profiles used in this study.

9:30

4aPA3. On the influence of the jet stream variations on infrasonic propagation. Maxwell B. Willis, Roger M. Waxler, and Claus Hetzer (National Ctr. for Physical Acoust., Univ. of MS, 145 Hill Dr., University, MS 38677, mwillis@go.olemiss.edu)

The jet stream is an eastward flowing wind jet seen around 10 km with its largest magnitude in the winter months, lessening in the summer. Its direction, while predominantly easterly, can vary from northeast to southeast on a day-to-day basis. It can produce ducting of infrasonic signals when its speed is of sufficient magnitude. Infrasonic signals were observed on a network of sensor arrays from a series of explosions detonated throughout the year. Each series consisted of 3 explosions, 1 series for each season of the year. Sensors were deployed in arrays of 4 sensors per array. The network consisted of 4 lines of arrays extending in the westerly, easterly, northeasterly, and southeasterly directions. These arrays were located at distances ranging from 20 km to 80 km from the source. The data from each array was analyzed to determine if infrasonic signals from the explosions were detected. A propagation model was produced for each event using a parabolic equation and an atmospheric profile corresponding to the time of each event. Correlations between the observations and the direction of the jet stream are investigated.

9:45

4aPA4. Sound propagation above a flat ground with random spatially varying properties. Didier Dragna and Philippe Blanc-Benon (Ctr. Acoustique, Laboratoire de Mécanique des Fluides et d'Acoustique, 36, Ave. Guy de Collongue, Ecully 69134, France, didier.dragna@ec-lyon.fr)

The acoustic properties of ground surfaces vary in space. Usually, these variations are not accounted for in predictions of outdoor sound propagation for two main reasons. First, they are not known as it is costly to obtain finely sampled spatial measurements of these ground properties. Second, there is no existing simple and quick-to-compute analytical solution for a ground with a spatially-varying admittance, contrary to the well-documented case of a ground with a homogeneous admittance. This imperfect knowledge of the ground characteristics leads to uncertainties about the sound pressure level. This paper aims at characterizing these uncertainties. Propagation of acoustic waves over the ground with a spatially-varying surface admittance is therefore considered. Using the diagram technique, the average Green's function is determined. An asymptotic expression in the form of a Weyl-Van der Pol formula is obtained at long range and at grazing incidence. Modifications of the reflection coefficient and of the surface wave contribution due to randomness are analyzed. Numerical simulations using a linearized Euler equations solver are then carried out. Comparison of the mean pressure obtained from ensemble-average over 200 realizations and from the analytical solution is performed. Finally, the mean intensity and intensity fluctuations are investigated.

10:00

4aPA5. Efficient modeling of long range impulse sound propagation in three dimensions. Matthias Cosnefroy, Sylvain Cheinet, Loic Ehrhardt (French-German Res. Inst. of Saint-Louis, 5 Rue du General Cassagnou, Saint-Louis 68300, France, matthias.cosnefroy@isl.eu), Daniel Juvé, and Didier Dragna (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Écully, France)

Sound characteristics close to the ground strongly depend on the atmospheric and ground properties in terms of amplitude, shape, and time of arrival. Time-domain numerical modeling is able to accurately account for outdoor sound propagation and is valuable to decipher the interactions between the refractive, scattering, and ground effects. It remains computationally challenging for three-dimensional long range simulations. A high-order parallel Finite-Difference Time-Domain solver with the moving frame approach is presented, featuring accurate time-domain impedance boundary conditions and very efficient convolutional perfectly matched layers to artificially truncate the computational domain. The design of the absorbing boundaries is based on a stability analysis of the time integration scheme and is shown to optimize the absorption properties for grazing waves. The model allows for accurate propagation of impulse sounds in 3D over several hundreds of meters and up to 1000 Hz using a personal laptop with a simulation duration of a few hours. The numerical predictions are compared to experimental measurements under different weather conditions, with a special focus on the sensitivity to the mean vertical wind profile and the ground properties.

10:15–10:30 Break

10:30

4aPA6. Performance of Green's function retrieval methods for various sound source distributions: Application to outdoor acoustic propagation. Max Denis (RDRL-CIE-S, U.S. Army Res. Lab., 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, Christian G. Reiff, and Deryck D. James (RDRL-CIE-S, U.S. Army Res. Lab., Adelphi, MD)

In this work, Green's function retrieval methods for an outdoor acoustic propagation channel are presented. Green's function retrieval methods by multidimensional deconvolution and crosscorrelation are compared for different source distributions. Of particular interest is the accuracy of the retrieved Green's function of an arbitrary sound source with that of an impulsive sound source. Also, the signal-to-noise ratio of both methods will be investigated. To this end, outdoor acoustic experiments are conducted in a clearing and wooded area in southern Maryland. Results will be obtained for various source-receiver ranges up to 400 m.

10:45

4aPA7. Atmospheric effects on acoustic vector sensing. Sandra L. Collier, Max Denis, Latasha Solomon, David Ligon, John Noble, W. C. Kirkpatrick Alberts, Leng Sim, Christian G. Reiff, and Deryck D. James (U.S. Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783-1197, sandra.l.collier4.civ@mail.mil)

Two-dimensional acoustic particle velocity and acoustic pressure, concurrent with atmospheric data, were collected during a series of field tests. It was found that for short propagation distances, the particle velocity field remained unsaturated (little variation in amplitude and phase) for higher frequencies than did the pressure field. Theoretical comparisons are made to examine the physical effects of the propagation environment on the particle velocity field statistics.

11:00

4aPA8. Numerical analysis on the effect of the sea surface roughness on the acoustic transmission loss in the atmosphere. Diego Turo (Mech. Eng., Catholic Univ. of America, The Catholic University of America, 620 Michigan Ave. NE, Washington, DC 20064, turo@cua.edu), Teresa J. Ryan (Engineering, East Carolina Univ., Greenville, NC), John Judge, and Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, Washington, DC)

Atmospheric sound propagation over the sea surface is affected by several factors including wind, temperature profile, and sea state. Preliminary numerical studies using a Crank-Nicolson parabolic equation solution approach have demonstrated that sea roughness introduces excess transmission loss. This excess loss induced by the sea roughness can be represented instead by a flat surface with a compensatory effective impedance. Calculation of transmission loss for the flat surface is computationally more efficient but requires the evaluation of the effective impedances for any given sea state. In this study, a 2-D finite-difference time-domain solver is used to determine the effective impedances for several sea states. The effect of sea roughness on transmission loss is then quantified by using the Crank-Nicolson parabolic equation.

11:15

4aPA9. Statistical characterization of sound propagation along near-vertical paths in a turbulent atmosphere. Vladimir E. Ostashev, D. Keith Wilson, and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu)

Most previous research on sound propagation in the atmosphere has focused on nearly horizontal propagation paths, with the sources and receivers close to the ground. Along such paths, refraction by wind and temperature gradients, and scattering of sound by small-scale turbulence, are known to be important and have been studied extensively during the past couple decades. Sound propagation from elevated sources is fundamentally different because the atmospheric motions are larger, more coherent, and driven by different types of instabilities. The corresponding research is needed to address the ability of ground-based acoustic microphone arrays to detect and track elevated sound sources, as well as the ability of elevated arrays to detect and localize ground-based sources. In this paper, near-vertical sound propagation through a turbulent atmosphere is studied using approaches and methods developed in wave propagation through random media. The statistical characteristics of sound signals for near-vertical propagation such as the log-amplitude and phase fluctuations, angle-of-arrival variance, transverse and longitudinal spatial coherence, and temporal and frequency coherence are calculated and analyzed. The results obtained are compared with those for near-horizontal sound propagation.

11:30

4aPA10. Nonlinear propagation of *N*-waves through kinematic turbulence: Statistics of peak overpressure and shock front steepness. Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Maria M. Karzova (Phys. Faculty, Moscow State University, Moscow, Russian Federation; LMFA - UMR CNRS 5509, Univ. Lyon, Ecole Centrale de Lyon, Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Sébastien Ollivier, Didier Dragna (LMFA - UMR CNRS 5509, Univ. Lyon, Ecole Centrale de Lyon, Ecully, France), Vera Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (LMFA - UMR CNRS 5509, Univ. Lyon, Ecole Centrale de Lyon, Ecully, France)

Nonlinear propagation of short *N*-waves (wavelength of 1.4 cm) through a turbulent layer with outer scale of about 16 cm was simulated based on the 2D parabolic KZK-type equation. A modified von Kármán spectrum was used to generate random fluctuations of wind velocity associated with the presence of turbulence. Statistics of peak overpressure and shock front steepness were analyzed for linear and various degrees of nonlinear effects in *N*-wave propagation. It was shown that in case of linear propagation, the turbulence mainly led to smearing the shock fronts and resulted in low probabilities of observing steepened shocks. At higher pressure level, nonlinear effects resulted in steepening of the shock fronts and hence counteracted the effects introduced by the turbulence. Due to nonlinear enhancement of focusing gain in random foci, up to twofold increase of the cumulative

probability of observing highly peaked waveforms was shown for nonlinear wave propagation comparing to the linear case. However, it was also shown that for stronger nonlinearities, saturation of the focusing gain led to saturation of the overpressure cumulative probabilities. [Work supported by RSF-17-72-10277 and by the Labex CeLyA of Université de Lyon, operated by the French National Research Agency (ANR-10-LABX-0060/ ANR-11-IDEX-0007).]

11:45

4aPA11. Deturbing supersonic signatures. William Doebler (The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, wfd5057@psu.edu) and Victor Sparrow (The Penn State Univ., University Park, PA)

The process of deturbing (removing effects of turbulence from) experimentally measured sonic boom pressure waveforms from aircraft would provide a baseline sonic boom signature. This baseline signature could be used to calculate perception metrics that may be part of a supersonic aircraft noise certification standard. The traditional deturbing process has been shown to remove turbulence reasonably well from experimentally measured N-wave sonic boom pressure waveforms whose tail shock generates the atmosphere's impulse response. N-wave booms such as that from Concorde were deemed unacceptable and prohibited for over land operations due to their startling nature and due to public annoyance. As a consequence, newly designed low-boom aircraft are being tailored for lower amplitude sonic booms that will not have any strong shocks. Since a tail shock is needed to generate the atmospheric impulse response for traditional deturbing, new deturbing techniques are needed. Two methods are presented using cross correlation averaging and frequency domain filtering for determining a baseline supersonic signature for computing sonic boom metrics. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

THURSDAY MORNING, 7 DECEMBER 2017 STUDIO 4, 9:00 A.M. TO 11:30 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Psychoacoustics of Speech Perception in Noise, Localization, and Frequency Selectivity

Eric Hoover, Chair
University of South Florida, 16458 Northdale Oaks Dr., Tampa, FL 33624

Contributed Papers

9:00

4aPP1. Auditory object formation from background noise improves speech perception. Maury Lander-Portnoy (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, landerpo@usc.edu) and Jason Zevin (Psychology, Univ. of Southern California, Los Angeles, CA)

Auditory object formation allows listeners to isolate a speech stream from a complex acoustic environment. While most studies focus on foregrounded speech, here we focus on whether listeners can form auditory objects from background noise to aid in speech perception. We use an open-set word recognition task with fluctuating speech-shaped-noise maskers, created by randomly interrupting the noise in each 125 ms window with a long (100 ms) or short (25 ms) silence. On each trial, three maskers were played, with the word occurring during the third. We previously showed that participants perform significantly better when the patterns repeat, compared to when the first two maskers differed from the third. Only the ordering of long or short within preceding patterns differs. To examine whether this advantage is due to auditory object formation, we leverage previous studies showing that inserted silences interfere with object formation. We insert 433 ms silences between patterns and observe defeat of the advantage we previously observed. We therefore attribute the advantage in the repeated masker condition to formation of an auditory object from background noise. Moving forward, we will manipulate other parameters affecting object formation and observe their affect on speech perception in noise.

9:15

4aPP2. Background noise interacts with spectral context effects during speech categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech perception is heavily influenced by surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects (SCEs)* that bias categorization of later sounds. For example, context sounds with a low-F1 bias produce more high-F1 responses to the target vowel and vice versa. SCEs have been studied for decades but only for speech perception in quiet. Background noise is expected to interfere with these effects, but the extent of this interference is unclear. On each trial, listeners heard a context sentence and target vowel both combined with speech-shaped noise. Sentences were filtered to add a +20-dB spectral peak at low-F1 (100–400 Hz) or high-F1 frequencies (550–850 Hz). Target vowels varied from /ɪ/ to /ɛ/. Background noise was unmodulated or sinusoidally modulated at 2/4/8 Hz at variable SNRs. In general, SCEs were larger at better SNRs and in unmodulated noise, but all SCEs were smaller than those reported for speech in quiet. Complex patterns of results were observed at different noise modulation rates, with likely ties to modulations in F1 frequency regions in the context sentence. In sum, background noise diminishes context effects in speech perception, but the extent depends on noise characteristics.

9:30

4aPP3. The effect of listener motion on localization of tones in a room. Eric J. Macaulay, Jack C. Magann, and William M. Hartmann (Phys. & Astronomy, Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, macaula5@msu.edu)

Listeners reported the perceived azimuths of 36 sine-tone sources in a room. For some trials, listener motion was forbidden and for others unstructured head and torso motion was encouraged with the listener remaining seated. Tone frequencies were 500, 1000, 2000, and 4000 Hz. Onsets and offsets were masked, leaving a steady-state duration of 7.6 s for exploration. It was found that the benefits of motion on source localization accuracy were clearly frequency dependent. Dynamic interaural cues were obtained from probe-microphone ear-canal recordings. Interaural cues were combined with head-tracker orientation data to predict perceived azimuths according to diverse hypotheses. Responses and interaural cues for trials without motion were used to predict a dynamic inferred location within the room for runs with motion and to identify stable inferences. Analysis of front/back confusions was conducted based on the regression between changes in interaural level differences and changes in head angle sampled at 76 instants. Except for 4000 Hz, the slope of the regression was significantly different depending on whether responses were in the front or back of the room. For interaural time difference, significance was limited to 500 and 1000 Hz. [Work supported by the AFOSR.]

9:45

4aPP4. Effects of listener's whole-body rotation and sound duration on sound localization accuracy. Akio Honda (Yamanashi-Eiwa College, 888 Yokone-machi, Kofu, Yamanashi 400-8555, Japan, honda@yamanashi-eiwa.ac.jp), Sayaka Tsunokake, Yôiti Suzuki, and Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Miyagi, Japan)

Listener's head movement is known to facilitate sound localization, which creates dynamic changes to the information input to both ears. For this study, we used a digitally controlled spinning chair to examine the effects of a listener's whole-body rotation and sound duration on sound localization accuracy. We measured their sound localization accuracy at locations from left 30 deg to right 30 deg with respect to the listener. Stimuli were 1/3-octave band noise burst (fc = 1 kHz, SPL = 65 dB) with duration of 50, 200, and 1000 ms. Each stimulus was presented from a loudspeaker in a circular array. The listener, sitting on the chair at the circle center, reported the position of the presented stimulus in chair-still (0 deg/s) and chair-rotation (10 deg/s) conditions. Results showed superior sound localization accuracy of chair-rotation condition to that of a chair-still condition. Moreover, a significant effect of sound duration was observed, but interaction of the test condition and the sound duration was not significant. Although the listener's head movement might influence the localization performance more for long stimuli than for short stimuli, results suggest that the effects of a listener's whole-body rotation were less influenced by the sound duration.

10:00

4aPP5. Diagnostic value of experimental parameters for measurement of localization ability. Angelique A. Scharine, Morgan C. Domanico, Ashley N. Foots, and Timothy J. Mermagen (Human Res. and Eng. Directorate, US Army Res. Lab., 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005-5425, angelique.a.scharine.civ@mail.mil)

This research examines the test parameters in two methods proposed by an ANSI/ASA working group for measurement of auditory localization with tactical communications and hearing protection systems (TCAPS). The localization task consisted of responding to pink noise bursts (0.250s and 7000s). Participants responded by pointing to the perceived source with a laser pointer mounted on an instrumented chair. 12 participants completed the study—half of the trials while wearing no headset and half while wearing an active Invisio X5 headset. Method 1 used four pairs of loudspeakers, separated by 8°, 10°, or 12°, and the listener was tested in two orientations (0° and 45°). Intended for use in untreated acoustic environments, Method 1 provides only a coarse estimate of localization accuracy and proportion of front/back errors. Method 2 used 36 equally spaced loudspeakers, a sound treated environment, and a chair instrumented with a laser pointer. Performance is compared as a function of separation between loudspeaker pairs in Method 1 and test method. The analyses include measures of bias, magnitude, and whether errors are due to poor acuity or front/back confusion. The utility of the proposed test configurations to assess the effects of headgear on auditory spatial awareness will be discussed.

10:15–10:30 Break

10:30

4aPP6. Distribution of spectral modulation transfer functions in a young, normal-hearing population. David A. Eddins, Eric Hoover, and Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu)

Common measures of auditory spectral shape perception include notch-noise masking methods to estimate auditory filter shape and width (i.e., equivalent rectangular bandwidth or ERB) and spectral modulation detection to estimate across-channel spectral processing. The notch noise masking method has been widely used and the distribution of ERBs in a young, normal hearing population is well established. The spectral modulation detection method is growing in popularity yet the distribution of spectral modulation transfer functions among young normal hearing people is unknown. Here, we report the distribution of spectral modulation detection thresholds for spectral modulation frequencies from 0.25 to 8 cycles/octave for 49 young, normal hearing listeners who inexperienced in performing psychoacoustic listening tasks. The carrier was Gaussian noise with a passband from 400 to 3200 Hz and −24 dB/octave rolloff outside the passband. The presentation level was 80 dB SPL. The results reveal the typical bandpass sensitivity to spectral modulation with a minimum near 2.0 cycles/octave. Thresholds at each frequency are tightly distributed across the population and normative values are computed including mean, median, and corresponding confidence intervals. These reference values can be very useful for future investigations involving normal controls or patient populations and when comparing among previously published datasets.

10:45

4aPP7. Detection of frequency glides in single tones and single formants: Are tones appropriate analogs of formants in psychophysical experiments? Michelle R. Molis and Amie Roten (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland HCS, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

Psychophysical studies of the limits on detection of frequency glides often have used single tones as simple, non-speech analogs of more complex speech stimuli. However, the perceptual equivalence of a single component falling at a nominal formant-peak frequency and a multi-component spectral resonance has not been firmly established. The distinction may be especially relevant when formant frequencies change over time. Detection thresholds for upward and downward frequency change were measured for single-component tonal glides and multi-component spectral resonances as a function of center frequency (500 or 1500 Hz) and duration (30 or 120 ms). The fundamental frequency of the formant stimuli was held steady at 104.5 Hz. The tones were presented at 15 dB SL and the formants were presented at approximately 15 dB SL for a 1/3-octave frequency region around the center frequency. Results suggest that thresholds for frequency glide detection depend on the complexity of the stimuli, as well as stimulus direction, center frequency, and duration. [Work supported by NIH/NIDCD.]

11:00

4aPP8. Comparison of scoring methods for spatial release from masking for speech based on analysis of psychometric function slope. Eric Hoover (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 16458 Northdale Oaks Dr., Tampa, FL 33624, erichoover@usf.edu), Anna Diedesch (Otolaryngol. Head and Neck Surgery, Oregon Health and Sci. Univ., Portland, OR), Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), and David A. Eddins (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

The ability to benefit from the spatial separation of talkers in a multi-talker environment is critical to speech communication. A headphone-based

test of spatial release from masking for speech (SR2) has been shown to be sensitive to differences between young, healthy listeners and listeners with various auditory deficits. Existing studies relied on a simplistic method to score SR2, comparing the number of correct responses in conditions of collocated and spatially-separated background talkers. Psychometric functions were fit to trial data compiled across multiple studies in order to evaluate if an alternative scoring method could improve diagnostic power. We hypothesized that differences in informational masking between collocated and spatially separated conditions would result in a difference in condition-specific slope estimates, thus leading to a relationship between scoring method and diagnostic power. Results showed a small but significant difference in slope between conditions, consistent with expectations, but the difference between scoring methods was not clinically significant. This information will be used to guide the development of a portable, rapid implementation of SR2 for clinical and research applications. [Funding provided by NIH R01 DC015051.]

11:15

4aPP9. Establishing the response of low frequency auditory filters. Menachem Rafaelof (National Inst. of Aerosp., M.S. 463, 100 Exploration Way, Hampton, VA 23681-2199, menachem.rafaelof@nasa.gov), Andrew Christian, Kevin P. Shepherd, Stephen A. Rizzi (NASA Langley Res. Ctr., Hampton, VA), and James Stephenson (US Army Aviation Development Directorate, Hampton, VA)

The response of auditory filters is central to frequency selectivity of sound by the human auditory system. This is true especially for realistic complex sounds that are often encountered in many applications such as modeling the audibility and annoyance of sound, voice recognition, noise cancelation, and the development of advanced hearing aid devices. The purpose of this study was to establish the response of low frequency (below 100 Hz) auditory filters. Two experiments were designed and executed; the first was to measure subjects' hearing threshold for pure tones (at 25, 31.5, 40, 50, 63, and 80 Hz), and the second was to measure the Psychophysical Tuning Curves (PTCs) at two signal frequencies (Fs= 40 and 63 Hz). Experiment 1 involved 36 subjects while experiment 2 used 20 subjects selected from experiment 1. Both experiments were based on a 3-down 1-up 3AFC adaptive staircase test procedure using either a variable level tone or variable level, narrow-band, noise masker. A summary of the results includes masked threshold data in the form of PTCs, the response of auditory filters, their distribution across subjects, and comparison with similar recently published data.

THURSDAY MORNING, 7 DECEMBER 2017 — STUDIO 7, 8:00 A.M. TO 10:50 A.M.

Session 4aSCa

Speech Communication: The Southern States: Social Factors and Language Variation I

Wendy Herd, Cochair
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Irina A. Shport, Cochair
English, Louisiana State University, 260-G Allen Hall, Baton Rouge, LA 70803

Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. Sociophonetic trends in studies of Southern English. Erik R. Thomas (English, North Carolina State Univ., Box 8105, Raleigh, NC 27695, erthomas@ncsu.edu)

Studies of Southern English have largely kept pace with advancements in sociophonetics in other parts of the world. They have expanded away from single-point studies of vowel nuclei to cover such topics as vowel-inherent spectral change, intonation, consonantal acoustics, and neurolinguistic coding of phones, especially with regard to ethnic identification. The ethnic diversity of the South, with African Americans, Latinos, enclaves of Asian Americans, and other groups such as Cajuns, facilitates growth in these new directions. Among the current linguistic trends of which sociophoneticians are staying abreast are the expansion of Latinos, social divergence within the African American population, and rapid dialect leveling among urban white Anglos. Integrating these developments with other advancements in sociolinguistics has proved more difficult. Empirical studies have incorporated various sociological methods with regard to social networks, but researchers are only now beginning to utilize them in conjunction with current phonetic techniques. There is also a need to examine wider arrays of phonetic variables in studies.

8:30

4aSCa2. Northern vowel features within the deep south: The new orleans yat dialect. Katie Carmichael (Virginia Tech, 407 Shanks Hall, 181 Turner St. NW, Blacksburg, VA 24061, katcarm@vt.edu)

Linguists have noted the distinctively Northern sound of the White, working class New Orleans English dialect locally referred to as Yat, though the features of this dialect have not been well documented phonetically. Vowel features attested include a split short-a system, BOUGHT raising, and Canadian raising of /ɑʊ/—common to Northeastern dialects but rare within the American South. In this study, I analyze the realization of these sounds within a corpus of interviews, reading passages, and word lists completed with 57 local Yat speakers, situating Yat within broader American English trends. Results indicate that in comparison to descriptions of these features in Northeastern dialects, the Yat split short-a system features slightly different phonetic triggers, BOUGHT is less raised and diphthongal, and Canadian raising appears to occur across the diphthong and not just in the nucleus. Thus, while these features do resemble those found in Northeastern dialects, there are some local distinctions. Moreover, BOUGHT raising and the split short-a system are in decline amongst younger speakers, while Canadian raising is a relatively recent innovation within the New Orleans area. Thus, Yat will likely remain a peculiar dialect within the South, despite its changing qualities.

8:55

4aSCa3. The emergence of a new pronunciation variant in a culturally changing appalachian community. Ewa Jacewicz and Robert A. Fox (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

The dialect spoken in the small southern Appalachian community in Western North Carolina, formally a part of the Inland South, represents the most archaic features of Southern English, including monophthongization of the diphthong /ai/ to [a:]. While Southerners produce the monophthong before voiced consonants ("prize") all across the South, pre-voiceless monophthongization ("price") is more restricted and still occurs in the Inland South. However, as the local culture is gradually shifting toward new mainstream sociocultural norms, so does the local dialect. To document the sound change in progress, this acoustic study examined the /ai/-monophthongization across several generations of local speakers ranging from 8 to 91 year olds, 58 males and 60 females, using a style shifting paradigm. Older generations produced the monophthong irrespective of speaking style and consonant context. Young adults introduced the diphthongal variant in pre-voiceless context. The actual departure from the monophthong occurred in children—more so in the girls—who created an intermediate, slightly diphthongized variant of /ai/, halfway between [a:] and /ai/. Bridging the two worlds, the children adjusted their pronunciation of this intermediate diphthong according to speaking style, also rejecting the archaic pre-voiceless monophthongization. Strategies for children's association of the spectral enhancement/reduction with communication context will be discussed.

9:20–9:35 Break

9:35

4aSCa4. The influence of regional identity on appalachian intonation. Paul E. Reed (Dept. of Communicative Disord., Univ. of Alabama, 909 Welsh Humanities Bldg., University of South Carolina, Columbia, SC 29208, reedpe@email.sc.edu)

In a study of intonation in Appalachian English (AE), Greene (2006) suggested that relative frequency and phonetic realization of pitch accents might reflect a stronger regional identity. The present study tests these observations, using rootedness—defined as one's orientation to place—as a means to describe observed intonational variation. Data were drawn from sociolinguistic interviews with 25 AE speakers from northeast Tennessee. In addition to the interview, every participant completed a Rootedness Metric, a psychometric survey designed to quantify place-based orientation. To consider the extent how AE speakers compared to the broader South, this cohort was also compared to a non-Appalachian Southern cohort. A section of speech from each speaker was labeled following MAE-ToBI conventions (Beckman *et al*., 2005). Pitch accent frequencies were totaled. Additionally, tonal peak alignment for the L + H* pitch accents was measured. Results indicate that AE speakers have a greater occurrence of L + H* pitch accents ($p<0.001$), and that the tonal peak occurs earlier in the syllable ($p<0.005$) compared to speakers of other Southern varieties. Within the AE cohort, both older and more rooted speakers have more frequent L + H* occurrence ($p<0.001$), while males ($p<0.01$), and more rooted speakers ($p<0.004$) have earlier peak alignment.

10:00

4aSCa5. An acoustic and phonological description of /z/-devoicing in Southern American English. Abby Walker, Amy Southall, and Rachel Hargrave (Virginia Tech, 409 Shanks Hall, 181 Turner St. NW, Blacksburg, VA 24060, ajwalker@vt.edu)

In this paper, we investigate whether z-devoicing is a feature of Southern American English. 37 students from around Virginia, half of whom identified as Southern, half of whom did not, were recorded completing a picture naming task and a reading task designed to elicit /z/-final tokens. They also answered questions about their orientation and attitudes toward their hometown, Appalachia, and the South in general. Participants' final /z/ tokens were automatically categorized as being [z] or [s] using the FAVE Aligner (Rosenfelder *et al*. 2011) and then acoustically analyzed in terms of duration, spectral energy measures, and voicing. The results of the categorical analysis and voicing analysis suggest that people who identify as Southern /z/-devoice at higher rates than those who do not, but only in prepausal environments. In fact, there appears to be significantly more voicing in Southern speech in pre-voiced environments. Additionally, within the Southern identifying participants, those who express more positive attitudes towards the South /z/-devoice more than those who do not. The results suggest that pre-pausal /z/-devoicing can be considered a socially meaningful feature of Southern American English.

10:25

4aSCa6. Prosodic and segmental cues to regional dialect variation in American English. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX) and Cynthia G. Clopper (Linguist, Ohio State Univ., 1961 Tuttle Park Pl., 108A Ohio Stadium East, Columbus, OH 43210, clopper.1@osu.edu)

Historically, most research on regional variation in American English (AE) examined lexical and segmental sources of variability. Focusing on prosodic variation, we have demonstrated significant effects of regional dialect on overall articulation rate, distributions of pauses, pitch accents, phrasal-boundary tone combinations, and syllable-to-syllable vowel and consonant duration variability (Clopper & Smiljanic, 2015, 2011). In this talk, we examine whether native AE listeners utilize prosodic and segmental cues to classify AE talkers by region and whether listener familiarity with different dialects affects classification. Using free classification with unmodified, monotone (flattened F0), and low-pass filtered (removed segmental information) stimuli, listeners from Ohio and Texas grouped 60 talkers based on perceived regional similarities. Listeners were more accurate in classifying talkers with the unmodified and monotone stimuli than with the low-pass filtered stimuli. There were no differences between the two listener groups in accuracy. Multidimensional scaling analyses indicated that listeners used gender and region as salient dimensions in classifying talkers in the unmodified and monotone conditions, but not in the low-pass filtered condition. The results suggest that segmental cues facilitated classification whereas intonation and global temporal cues on their own did not. The relationship between acoustic-phonetic features and classification patterns will be discussed.

THURSDAY MORNING, 7 DECEMBER 2017 — STUDIOS FOYER, 11:00 A.M. TO 12:00 NOON

Session 4aSCb

Speech Communication: The Southern States: Social Factors and Language Variation II (Poster Session)

Wendy Herd, Cochair
Mississippi State University, 2004 Lee Hall, Drawer E, Mississippi State, MS 39762

Irina A. Shport, Cochair
English, Louisiana State University, 260-G Allen Hall, Baton Rouge, LA 70803

All posters will be on display and all authors will be at their posters from 11:00 a.m. to 12:00 noon.

Contributed Papers

4aSCb1. Incipient /ay/-Raising in Baton Rouge. Kelly Berkson (Linguistics, Indiana Univ., 1020 E. Kirkwood Ave., Dept. of Linguist - Ballantine 844, Bloomington, IN 47405, kberkson@indiana.edu) and Wendy Herd (MS State Univ., MS State, MS)

Canadian raising targeting the /ay/ diphthong has been reported for a number of dialects of U.S. English. In the United States and elsewhere, however, the incipient—or purely phonetic—stage of diphthong raising, wherein it is triggered only by consonants that are phonetically voiceless, has been notoriously difficult to capture. In such a stage, raising is expected before the /t/ in *cite* (/saɪt/ → [sʌɪt]) but not before the flapped-/t/ in *citing* (/saɪtɪŋ/ → [saɪɾɪŋ]). Berkson, Davis, and Strickler (in press) recently discovered incipient phonetic raising in northeastern Indiana, however, and have suggested that extremely short diphthongs immediately preceding a primary stress (as in *citátion, psychólogy*) may be the very first to undergo raising. We investigate this hypothesis with data from Louisiana. While Yat English spoken near New Orleans has been reported as an /aw/-raising dialect (Carmichael, 2012), we also find /ay/-raising in New Orleans. Furthermore, the question of whether raising is present in Baton Rouge remains unanswered. Data presented here reveal that incipient phonetic raising is present in some—but not all—talkers from Baton Rouge. We consider the variation in Baton Rouge /ay/-raising in the context of gender differences and of /ay/-monophthongization typical in Southern varieties of English.

4aSCb2. Perception of phonetic imitation of Southern American English by Midwesterners. Cynthia G. Clopper and Ellen Dossey (Ohio State Univ., 1961 Tuttle Park Pl., 108A Ohio Stadium East, Columbus, OH 43210, clopper.1@osu.edu)

Phonetic imitation has been observed across regional dialects in English, French, and Mandarin. This cross-dialect imitation is affected by social factors, including dialect prestige and social salience of the linguistic variables. The goal of the current study was to explore the perception of phonetic imitation of Southern American English by native speakers of Midwestern American English in a word shadowing task. The magnitude of the Midwesterners' phonetic imitation was assessed using an AXB perceptual discrimination task, in which a different group of Midwestern listeners was asked to identify either a baseline read token or a shadowed token as more similar to the model token. The results of the AXB task revealed evidence of overall imitation, consistent with acoustic measures of imitation in word durations, vowel formant frequencies, and vowel formant trajectories. Moreover, perceived imitation was weaker for shadowers who were told where the model talker was from (i.e., Louisville, Kentucky) than for shadowers who were not provided with any information about the model talker, suggesting that explicit information about the model talker's dialect background can shape the magnitude of cross-dialect phonetic imitation. The specificity of this cross-dialect phonetic imitation across acoustic domains and vowel variables will be discussed.

4aSCb3. Acoustic correlates of perceived southernness ratings. Kaylynn Gunter, Charlotte Vaughn, and Tyler Kendall (Linguist, Univ. of Oregon, 1585 E. 13th Ave., Eugene, OR 97403, kgunter@uoregon.edu)

The Southern U.S. dialect and the Southern Vowel Shift (SVS), in particular, have been the subject of extensive research (e.g., Feagin, 1986; Labov, 1991; Fridland & Kendall, 2015), though there is limited work examining what acoustic cues trigger listeners to judge a speaker as sounding southern (cf. Fridland, Bartlett, & Kreuz, 2004; Allbritten, 2011). Fridland & Kendall (2012), and others, have used the Euclidean distance (ED) between the front vowels /e/ and /ɛ/ as a gradient metric of speakers' degree of SVS shiftedness. While this was demonstrated to be a useful diagnostic in production, it has not been tested against listeners' percepts of speakers' southerness. This study asks: are listeners' perceptions of southernness predicted by a speaker's /e/-/ɛ/ ED, or other such measures? To test this question, we presented listeners with isolated words from both southern and western speakers. Listeners rated words on a 1–9 scale of how southern they sound. We assess whether southernness ratings are predicted by speakers' /e/-/ɛ/ ED and other acoustic correlates. We examine both the perception of vowels implicated in the SVS, in addition to those less associated with the shift. Results shed light on the acoustic correlates of perceived southernness in vowel production.

4aSCb4. Southern stops: Phonation type differences in Mississippi. Wendy Herd (Mississippi State Univ., 2004 Lee Hall, Drawer E, MS 39762, wherd@english.msstate.edu)

While the voicing contrast between American English word-initial stops is often described as relatively uniform across speakers (e.g., voiced segments are produced with short positive VOTs while voiceless segments are produced with long positive VOTs), considerable sociophonetic variation exists. The current study investigates variation in the VOT of voiced and voiceless word-initial stops in *pot, bot, tot, dot, cot, got* produced in isolation and in carrier sentences. Participants included 40 native English speakers from Mississippi grouped according to their self-identified gender and ethnicity. As previously reported, African American speakers produced significantly more voiced stops with negative VOTs and more fully voiced closures preceding voiced stops than Caucasian American speakers. While speakers did not differ in their production of voiceless VOT when words were read in isolation, African American speakers maintained closure voicing preceding voiceless stops far more often than Caucasian American speakers. No gender differences were found. These data suggest this voicing variation is due to robust dialectal differences. The possibility that speakers who exhibit more prevoicing and more fully voiced closures are prolonging closure voicing through laryngeal lowering or nasal venting is explored through the examination of pitch and intensity trajectories within the closure and within the following vowel.

4aSCb5. Pitch accents in read speech: Black and White southern women. Yolanda F. Holt and Balaji Rangarathnam (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu)

Prosodic variation between African American English and General American English has been attested to in numerous works, yet few studies have collected measures of F_0 in African American English and fewer have examined F_0 beyond the word level. Additionally, the analysis of prosodic variation in regional dialects of American English is not well studied. F_0 movement at the level of the Intonational Phrase (IP) is known to convey both local and global information. Research on F_0 movement in General American English has analyzed combinations of H(igh) and (L)ow pitch accents as categorical markers of prosodic alignment to the segmental string. Understanding the alignment of F_0 contours provides key information on phonetic realization and phonological alignment in the creation of intonational categories. This pilot data explores the interaction of F_0, vowel duration and word duration of prenuclear and nuclear pitch accents in the read speech of Black and White southern women. This study seeks to determine if group differences exists in the expression of pitch accents between the regionally defined socio-ethnic dialects used by the two groups. Results will be discussed in terms of dialect variation.

4aSCb6. Comparing lexical decision reaction times and error rates for Southern and British English. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu)

The English spoken in the southern states of the US as well as the "received pronunciation" of British English are two varieties that the majority of students at the University of Southern California are relatively unfamiliar with. Some of the popular stereotypes surrounding the two groups of speakers, however, diverge: while it has been shown that certain listeners consistently down-grade Southern speakers on positive attributes like intelligence and competence, no such effect is observed for British speakers. This study aimed to investigate the impact of negative stereotypes on lexical decision reaction times and error rates. A list containing high frequency words, low frequency words and non-words was constructed and a trained voice actor recorded the stimuli in both British and Southern English. Participants heard half of stimuli in each variety. Participants tended to incorrectly reject more real words in Southern English than British. These errors occurred in trials with relatively short reaction times. British English stimuli tended to have slower reaction times. The implications of these findings in light of other experiments comparing lexical decision in these and other varieties of English are discussed.

4aSCb7. Acoustically quantifying /ai/ monophthongization in four southern dialect regions. Rachel M. Olsen, Michael Olsen, and Margaret E. Renwick (Linguist, Univ. of Georgia, 142 Gilbert Hall, University of Georgia, Athens, GA 30602-6205, rmm75992@uga.edu)

The *Atlas of North American English* describes four southern U.S. dialect areas: Inland South (IS; interior Appalachia), Texas South (TS; around Dallas), Florida (FL), and South (S; remainder of Southern US). These areas are distinguished by degree of /ai/ monophthongization, the Southern Vowel Shift's (SVS) triggering feature (Labov *et al.*, 2006). IS is argued to be most advanced in the SVS, and /ai/ weakens in all phonetic environments. While not as advanced generally, TS also features all-environment weakening. In S, /ai/ weakens only in syllable-final and pre-voiced conditions, and in FL it remains diphthongal. This study uses the *Digital Archive of Southern Speech (DASS)* to acoustically measure /ai/ weakening in each region. Speech was force-aligned and F1 and F2 values collected at five time points. Related work on F2 formant angle in *DASS* speech has suggested /ai/ weakens most in IS (Renwick and Stanley, 2017). Here, trajectory length (TL) measurement (Fox and Jacewicz, 2009) corroborates evidence for regional effects on diphthong dynamics, where longer TL corresponds to more diphthongal production. Mixed modeling suggests that IS and TS speakers monophthongize /ai/ more than those in S and FL. The effects of phonetic environment, race, and gender are also explored.

4aSCb8. Dynamic trajectories of tense vs. lax vowels in the American South. Margaret E. Renwick (Dept. of Linguist, Univ. of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu)

Regional variation in American English speech is often described in terms of vowel shifts, in which one or more sounds "move" within the X,Y plane of height and backness captured by vowels' first and second formant values. Such shifts indicate which sounds are converging or diverging, providing predictions for language change and variation. Static characterizations of shifting, with a single pair of F1/F2 values taken near the vowels' midpoint or intensity peak, provide an approximation of vowels' relations, but fail to capture acoustic information during the rest of the vowels' time course. In Southern dialects, where front tense and lax vowels may "swap places," static measures show strong overlap between these pairs and thus predict mergers between, e.g., *beet-bit*, *bait-bet*; instead, these mergers' clear lack of occurrence indicates the inadequacy of static methods. Using data from 64 semi-spontaneous linguistic interviews in the Digital Archive of Southern Speech, we model the shape and movement of vowels' dynamic trajectories. Formant values taken at five locations show that tense and lax vowels have distinct beginning and endpoints, though they pass through shared acoustic space near their temporal midpoints. Sociolinguistic factors affecting trajectories' length and shape are investigated using Generalized Additive Mixed Models.

4aSCb9. A preliminary examination of the gender role in back-vowel fronting in Central Louisiana. Irina A. Shport (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu)

In addition to the Southern Vowel Shift that involves /ɛ ɪ/ fronting, high back vowels also tend to be fronted in white Southern U.S. speech (Labov, Ash, & Boberg, 2006; Thomas, 2001). The tense /u/ is the first to shift in the back-vowel system, followed by the lax /ʊ/. Gender effect on the degree of fronting has been reported for one vowel but not the other: Male speakers lead in /u/fronting, whereas /ʊ/ fronting occurs more or less uniformly across speakers of different genders (Clopper, Pisoni, & de Jong; Fridland, 2001). This study provides a preliminary examination of the role of gender in the relative degree of back vowel fronting in young adult speakers from Central Louisiana. They were recorded producing words with target vowels in a variety of tasks: word list and passage reading, sentence creation, and informal conversation. The formant values were analyzed with a reference to other vowels in each speaker and with a reference to average values for Southern U.S. English reported in previous research. The data are discussed in the context of variation in the South, adding Central Louisiana to the linguistic map.

4aSCb10. Flapping before a stressed vowel: The case of *whatever*. Irina A. Shport, Gregory Johnson (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu), and Wendy Herd (English, State Univ., MS)

In English, word-internal intervocalic alveolar stops are predominantly flapped when preceding an unstressed vowel (*water, charity*) and optionally flapped at word boundaries preceding unstressed and stressed vowels (*that is, private airplane*). In this study, we show that /t/ is flapped in *whatever* although it is word-internal and precedes a stressed vowel. The data were elicited in a sentence reading task, with four speakers of Appalachian English. The duration of /t/ and the acoustic correlates of stress were examined. A comparison of vowel duration and amplitude patterns in *whatever* versus *everwhat* (both words are relative pronouns in free relative clauses in this dialect, $N = 699$) showed that the second syllable is stressed in *whatever*. A comparison of /t/ durations showed no significant differences among *whatever*, *watermelons, waterlilies, water buffalo* ($N = 533$, $M = 30.1$ ms). These results may be interpreted as: (a) *whatever* is an exception to the word-internal flapping environment, or (b) the word-internal flapping environment must be modified to include preceding stressed vowels at morpheme boundaries, or (c) *whatever* consists of two phonological words and falls within the word-final flapping environment. Prosodic and syntactic analyses of free relative clauses consistent with the last interpretation are discussed.

4aSCb11. Social and linguistic influences on the phonetic imitation of Southern American English vowels. Ellen Dossey (The Ohio State Univ., 108A Ohio Stadium East, 1961 Tuttle Park Pl., Columbus, OH 43210, dossey.1@osu.edu)

This study investigated the effect of regional biases on phonetic imitation, specifically exploring imitation of Southern American English by Midwesterners in a lexical shadowing task. To manipulate the regional biases at play, participants received different information about the model talker's regional origins. They were either told she was from the Midwestern city of Columbus, Ohio; the Southern locations of Eastern Kentucky or Savannah, Georgia; or nothing at all. After shadowing, participants completed a survey about their perceptions of the talker and her supposed region of origin. Vowel formant frequencies and trajectories were analyzed for imitation. Significant imitation was observed, although not all vowels were imitated equally. The survey revealed that the Southern locations differed in the biases associated with them, and that the Midwest was more socially desirable than the South. However, the survey responses were not predictive of imitative behavior. The vowel selectivity in imitation may be due to linguistic factors, such as the baseline distance between the model talker's and participants' vowels, or social factors such as the relative level of social stigma associated with the vowel variants. The results of this study suggest that imitation may be partly automatic but also influenced by certain language-external social information.

4aSCb12. Vowel acoustic characteristics of Southern White English produced by speakers from New Orleans area, Louisiana. Hyunju Chung and Lauren A. de Mahy (Dept. of Commun. Sci. and Disord., Louisiana State Univ., 81 Hatcher Hall, Field House Dr., Baton Rouge, LA 70803, hchung@lsu.edu)

This study aims to characterize vowels produced by speakers from the New Orleans area, Louisiana. The state of Louisiana is well known for its various dialects spoken across the regions. Despite the possible methodological issues and challenges researchers face for studying speech and language of young children (e.g., Oetting, & McDonald, 2002), vowel characteristics of different dialects, especially those of the New Orleans area, have not been well investigated. In the current study, vowels produced by speakers of Southern White English of New Orleans will be compared to those of two other regions of Louisiana. Participants included a total of 30 female adults, ranging in age from 18 to 25. There were 10 participants in each of the New Orleans, other Southern, and Northern Louisiana area. Each participant was asked to produce English words containing 11 different monophthongs three times in a randomized order. Vowel duration, the first three formant frequencies at the vowel midpoint, and formant trajectories were measured and analyzed. Acoustic differences in vowels are expected among speakers of three different regions. The outcome of the current study will provide normative data necessary for evaluating children's vowel articulation skill of diverse dialectal backgrounds.

Session 4aSP

Signal Processing in Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Signal Processing in Acoustics Metamaterials

Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab., 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Matthew Guild, Cochair
University of Texas at Austin, Austin, TX 78712

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Computational imaging with a frequency-diverse metasurface aperture. David Smith (Duke Univ., Box 90291, Durham, NC 27708, drsmith@ee.duke.edu)

We demonstrate image formation with radio frequency (RF) waves using a frequency-diverse metasurface antenna. The metasurface antenna consists of a planar cavity—formed from a double-layer, copper-clad printed circuit board—fed from one side by an RF source, and with a series of irises patterned into the opposite side. At any particular frequency, the cavity mode excited by the feed in turn excites the irises, each of which radiates as a magnetic dipole element. The cavity walls (fabricated using vias patterned into the circuit board) are irregular, so that the excited cavity modes exhibit considerable variation as a function of frequency. Thus, the radiated fields from the cavity-backed metasurface tend to have a series of randomly directed lobes, the number and directions of which vary significantly as a function of frequency. These radiation patterns can be used to form measurements of a scene, with the scene then estimated using computational imaging techniques. We have applied the metasurface aperture in a prototype imaging system capable of acquiring high resolution images of human-scale targets. The metasurface aperture design and reconstruction methods are general, and can be easily adapted for acoustic imaging.

8:25

4aSP2. Non-reciprocal acoustic metamaterials for full-duplex communications and imaging. Andrea Alu, Curtis Wiederhold, Li Quan, and Dimitrios Sounas (The Univ. of Texas at Austin, 1616 Guadalupe St. UTA 7.215, Austin, TX 78701, alu@mail.utexas.edu)

In this talk, we discuss our recent efforts in the area of non-reciprocal metamaterials based on moving media or spatiotemporally modulated acoustic elements, aimed at realizing efficient and compact isolation and circulation in acoustic devices. We discuss the relevance of these concepts for new-generation acoustic devices that realize isolation in guided and radiating structures, with relevance for acoustic imaging applications, including ultrasound and sonar technology, and for sound communications. Non-reciprocal responses and built-in isolation and circulation in waveguides and in radiating elements offers the opportunity of translating the advantages recently enabled by non-reciprocal components in electromagnetics, based on full-duplex signal transmission, to acoustic technology.

8:45

4aSP3. High speed acoustic communication with orbital angular momentum multiplexing. Chengzhi Shi, Marc Dubois, Yuan Wang, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, CA 94720, chengzhi.shi@berkeley.edu)

Acoustic communication is critical for underwater application such as deep ocean scientific explorations, off-shore industrial controls, and ocean environment monitoring. This is because other techniques using electromagnetic waves such as RF communications are difficult for underwater applications due to the strong absorption of water in such a frequency. Optical communication, on another hand, suffers from the light scattering from micro-particles or marine life making long range underwater optical communication very challenging. Therefore, using acoustic waves to transmit information is currently the dominate technique for underwater applications including data collection and remote control of off-shore benthic stations. However, the low frequency bandwidth available for acoustic communication limits the data transmission rate and information capacity or content. We propose and experimentally demonstrate here a new approach using the orbital angular momentum (OAM) of acoustic vortex beams which provides a new and independent channel that enhances the data transmission rate by eight fold. The OAM multiplexing method demonstrated here will impact significantly on future underwater communications.

9:05

4aSP4. Metamaterial-based passive phased arrays: Resolution, losses, and characterization. Likun Zhang, Joel Mobley (Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38677, zhang@olemiss.edu), Xue Jiang (Nanjing Univ., Nanjing, China), and Yong Li (Tongji Univ., Shanghai, China)

Metamaterial and metasurface-based passive phased arrays provide novel means for the manipulation of acoustic fields. As passive elements, metasurfaces can provide local phase delays which enable the steering of sound fields and facilitate extraordinary wave phenomena. Some aspects that affect the performance of these passive elements include the spatial resolution/aliasing effect, the component resonances that shape the transmission spectra, and the thermoviscous losses that limit the transmission efficiency [X. Jiang, L. Yong, and L. Zhang, *J. Acoust. Soc. Am.* **141**(4), EL363–368, April 2017]. Efficient acoustic field characterization is essential for investigating these issues with the ultimate aim of optimizing the geometrical parameters of the structure. This talk will address some details of the physics of metamaterial-based passive arrays and the broadband approach for assessing their transmission properties, with the aim of improving the design cycle and exploring novel uses.

9:25

4aSP5. A diffraction based acoustic single pixel imager. Jeffrey S. Rogers, Charles Rohde, David Smith (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil), Christina J. Naify (Jet Propulsion Lab., Pasadena, CA), and Gregory Orris (Acoust. Div., Naval Res. Lab, Washington, DC)

We present a method for localizing an acoustic source with a single, omni-directional receiver paired with shaped aperture screens that allow for a spatially diverse set of measurements. Traditionally this is accomplished by patterning the screen with a series of sub-wavelength openings that allow the acoustic transmission an otherwise sound opaque material. Here we consider screens that have openings on the order of an acoustic wavelength or larger and incorporate a diffraction model into the single pixel imaging framework to account for these larger openings. The method is demonstrated on experimental data taken in air and an analysis of the error as a function of receiver position is presented. [This work was supported by ONR.]

9:45–10:00 Break

10:00

4aSP6. Tailoring the flow of acoustic waves by architectured metamaterials. Nicholas X. Fang (MechE, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, nicfang@mit.edu)

Today, sound is an indispensable component in numerous industrial and consumer products, such as musical instruments, cars, building technology, medical diagnostics, and many others. Acoustic characteristics are among their most important properties, greatly influencing their function and our society at large. Recent development of acoustic metamaterials opens a door to an unprecedented large design space for acoustic properties such as negative bulk modulus, negative density, and refractive index. These novel concept expands paves the way for the design of a new class of acoustic materials and devices with great promise for diverse applications, such as broadband noise insulation, sub-wavelength imaging and acoustic cloak from sonar detection. In this invited talk, I will present our development of advanced design and micro/nanofabrication techniques, to enable exploration architectured meta structures for acoustic waves. These structures show promise on focusing and rerouting ultrasound through broadband metamaterials. As example, our study on the sound absorption of thin composite aerogel foams using a bimodal porous structure predicts a possible route to perfect thin film absorber by increasing the amount of epoxy resin. In a second case, bifunctional acoustic lens can be implemented in practice with subwavelength unit cells exhibiting effective anisotropic parameters. Lastly I will report our study on a prototype hydraulic hydrogel actuators with excellent optical and sonic transparency.

10:20

4aSP7. Acoustic characterization of silica aerogel clamped plates for perfect absorption purpose. Alan Geslain (ISAT, Dr. EA1859, Universite Bourgogne Franche Comte, Nevers, France), Vicente Romero-Garcìa, Jean-Philippe Groby (LAUM UMR CNRS 6613, Universite du Maine, Le Mans cedex 9, France), Francisco Cervera, and Jose Sanchez-Dehesa (Electron. Dept., Universitat Politecnica de Valencia, Camino de vera s.n., Valencia 46022, Spain, jsdehesa@upv.es)

Silica aerogel has been widely studied as bulk material for its extremely low density and thermal conductivity. Plates or membranes made of this extremely soft materials exhibits interesting properties for sound absorption. A novel signal processing method for the characterization of an acoustic metamaterial made of silica aerogel clamped plates is presented. The acoustic impedance of a silica aerogel clamped plate is derived from the elastic theory for the flexural waves, while the transfer matrix method is used to model reflection and transmission coefficients of a single plate. Experimental results are obtained by using an acoustic impedance tube. The difference between the measured and modeled reflexion and transmission coefficients is minimized under constraints to recover the acoustic parameters of the silica aerogel plate. Once the properties of the silica aerogel plate are obtained, the perfect absorption condition is derived by studying the reflection coefficient of a aerogel plate rigidly backed with a cavity in the complex frequency domain. Reflection measurements with a varying cavity length from 1 mm to 65 mm are performed to valid the perfect absorption condition. It is found that the use of silica aerogel plate exhibit perfect absorption condition for several configurations.

10:40

4aSP8. Constant amplitude sound waves in non-Hermitian metamaterials. Etienne Rivet (Inst. of Elec. Eng., EPFL, Lausanne, Vaud, Switzerland), Andre Brandstötter (Inst. for Theor. Phys., TU Vienna, Vienna, Austria), Konstantinos Makris (Phys. Dept., Univ. of Crete, Heraklion, Greece), Stefan Rotter (Inst. for Theor. Phys., TU Vienna, Vienna, Austria), Herve Lissek (Inst. of Elec. Eng., EPFL, Lausanne, Switzerland), and Romain Fleury (Inst. of Elec. Eng., EPFL, 1 University Station C0803, Austin, TX 78712, romain.fleury@epfl.ch)

We investigate the possibility for acoustic waves to propagate with a constant amplitude in disordered media. We find that this remarkable property is possible if one adds a tailored distribution of gain and loss on top of the disorder, making the medium non-Hermitian. We present the theory of constant-amplitude acoustic waves in both cases of continuous and discrete media, and provide an experimental demonstration in a metamaterial at audible frequencies.

11:00

4aSP9. Breaking the fundamental limit of space-coiling metamaterial: Toward simultaneous phase and amplitude modulation. Reza Ghaffarivardavagh, Jacob Nikolajczyk (Mech. Eng., Boston Univ., Boston, MA), Stephan Anderson (Radiology, Boston Univ. Medical Ctr., Boston, MA), and Xin Zhang (Mech. Eng., Boston Univ., One Silber Way, Boston, MA 02215, xinz@bu.edu)

Acoustic metasurfaces represent a family of planar, wavefront-shaping devices garnering increasing attention due to their capacity for novel acoustic wave manipulation. Despite the successful demonstration of phase engineering using metasurfaces, amplitude modulation remains overlooked. This work explores the feasibility of simultaneous phase and amplitude modulation using space-coiling metamaterials. In the case of conventional space-coiling metamaterials, we observed a fundamental bound on the transmission coefficient, which precludes full wavefront manipulation. Herein, we present a novel class of metasurfaces featuring a modified space-coiling structure and enabling full acoustic control with simultaneous phase and amplitude modulation. The functionality of this class of metasurfaces, featuring a gradient in channel spacing, has been theoretically and numerically investigated and an equivalent model simplifying the structural behavior is presented. Furthermore, a metasurface featuring this novel geometry has been designed and its functionality in modifying acoustic radiation patterns simulated. The class of acoustic metasurface demonstrated in this work provides a new design methodology enabling complete acoustic wave manipulation, which may find utility in a range of applications including biomedical imaging, acoustic communication and non-destructive testing.

11:20

4aSP10. Deconvolution methods to obtain the impulse response of acoustic metamaterial samples of finite extent. Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, sprattkyle@gmail.com), Colby W. Cushing, Kevin M. Lee, Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Xiaoshi Su, and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

One complication of characterizing the response of acoustic metamaterials is that models often assume the medium is of infinite extent. On the other hand, material property measurement can only be done with samples of finite size. The result is that acoustic field measurements include the effects of edge diffraction and scattering from fixtures. This work presents deconvolution methods to extract the frequency-dependent reflection and transmission behavior of acoustic metamaterial samples. Measurements using broadband chirp signals are obtained and subsequently post-processed using deconvolution techniques to obtain high-time-resolution impulse responses. Additionally, the chirp signals can be modified to compensate for the frequency response of the transducers being used. Examples will be shown for the transmission behavior through a two-dimensional pentamode gradient index lens and the reflection response of a flat square plate. [Work supported by ONR.]

11:40

4aSP11. The use of acoustic resonators for characterization of underwater acoustic metamaterials. Preston S. Wilson and Michael R. Haberman (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

The characterization of acoustic metamaterials used in underwater applications at low frequencies (below a few kilohertz) is complicated by the associated long wavelengths (1 m and greater), which render free field methods impractical for a range of frequencies of significant interest. One approach used to address this issue is the water-filled acoustic resonator. The key benefit is that a resonator tube with a length of as little as one half wavelength can be used for material characterization, compared to free-field techniques, which require a spatial extent of several wavelengths in two or more dimensions. Another benefit is that sample size can be orders of magnitude smaller when using a resonator. These benefits are enabling, but come at the expense of designing a system to overcome the fact that practical tube materials such as steel or glass cannot be considered rigid when water-filled. Another caveat is that an appropriate model is required to relate the observed resonance frequencies to the acoustic properties of interest. An overview of resonator-related measurement techniques will be presented, along with recent efforts to apply these techniques to the measurement of underwater acoustic metamaterial properties. [Work supported by ONR.]

Session 4aUW

Underwater Acoustics: Underwater Soundscapes and Noise: Measurement and Abatement

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, MSN 1G5, Fairfax, VA 22030

Aleksander Klauson, Cochair
Civil Engineering and Architecture, Tallinn University of Technology, Ehitajate tee 5, Tallinn 19086, Estonia

Contributed Papers

8:00

4aUW1. Low frequency ambient noise modeling in the North Pacific: Simulation versus experiment. Mehdi Farrokhrooz and Kathleen E. Wage (George Mason Univ., 4211 Ridge Top Rd., Apt. 2114, Fairfax, VA 22030, farrokhrooz1@gmail.com)

Farrokhrooz *et al*. [*J. Acoust. Soc. Am*., 2017] measured the seasonal variations of ambient noise in the 40–50 Hz band using long vertical arrays deployed during SPICEX, a 2004–2005 experiment in the northeastern Pacific. This talk compares simulations of low frequency noise from distant sources with the SPICEX data. The dominant sources of distant noise in the 40–50 Hz band are ships and wind. Noise from surface shipping and wind sources is transferred into the deep sound channel by several mechanisms [Dashen and Munk, J. Acoust. Soc. Am. 1984] and propagates long distances. The simulations use shipping density maps from exactEarth, Ltd., and measured wind speeds from European Centre for Medium-Range Weather Forecasts (ECMWF). A parabolic equation model that includes internal wave effects is used to generate Monte Carlo trials for different realizations of the shipping and wind environment. Results indicate that the seasonal variations in the SPICEX data are dominated by distant wind sources, including storms and persistent high latitude winds during the winter season. This comparison of simulation and experiment supports Bannister's conclusions [J. Acoust. Soc. Am., 1986] regarding the role of high latitude winds in generating low frequency ambient noise. [Work sponsored by ONR.]

8:15

4aUW2. Assessment of the proportion of anthropogenic underwater noise levels in passive acoustic monitoring. Mirko Mustonen, Aleksander Klauson (Dept. of Civil Eng. and Architecture, Tallinn Univ. of Technology, Ehitajate tee 5, Tallinn 19086, Estonia, mirko.mustonen@gmail.com), Mihkel Tommingas, and Julia Berdnikova (Thomas Johann Seebeck Dept. of Electronics, Tallinn Univ. of Tecnology, Tallinn, Estonia)

The marine environment faces a pressure from the increasing shipping intensity in the form of rising levels of continuous anthropogenic underwater noise. Current underwater noise monitoring guidelines advise to measure the long-term trends in the overall noise levels in the frequency bands where ship noise is most prevalent. However, the natural noise is omnipresent and prolongs the time period required for the detection of statistically significant trends in the overall noise. The monitoring efficiency can be improved by finding the proportions of the anthropogenic and the natural noise levels and by measuring the changes in the proportions over time. These proportions can be found by differentiating between the two types of ambient noise in the recordings according to both proximity of the ships and the variability of the environmental conditions. This is achieved by using the AIS ship traffic data along with the ship noise detection algorithms. The AIS data enable to determine the position of ships around a noise monitoring location and calibrate the ship noise detection algorithms. The results and the methods are presented for the passive acoustic monitoring in the Baltic Sea and applicability of the described methods are discussed.

8:30

4aUW3. The marine soundscape off the Isle of Mull in Scotland's Inner Hebrides. Adele Roland, Kathleen E. Wage, and E. C. M. Parsons (George Mason Univ., 4400 University Dr., Fairfax, VA 22030-4444, roland.adele@gmail.com)

The waters around the Isle of Mull, Scotland are home to minke whales, basking sharks, common and bottlenose dolphins, harbor porpoises, and many fish and marine invertebrates. This talk describes the soundscape of this region, measured as part of a study on minke whale (*Balaenoptera acutorostrata*) acoustic habitat use, conducted June–September 2016. The study area consists of shallow (<100 m depth) shelf waters between and around the Islands of Mull, Coll, and Muck. Measurements were made with a Soundtrap 300 from multiple points over the study area (~560 km^2) at a sampling rate of 288k Hz. Recordings were taken at a depth of ~30 m from a whale watching vessel when it was stopped (usually observing marine wildlife). The study includes 131 recordings with durations of 5–40 min. Spectral density varies with location and time over the study. Temporal and regional variation and the identity of various sounds are analyzed. Most of the acoustic energy in the region is concentrated below 50 kHz. The soundscape is dominated by wave noise, snapping shrimp, and distant dredging. Local areas or times may be dominated by transient biological sounds, e.g., the calls of common dolphins. Regular but infrequent ferry passage temporarily alters the base soundscape.

8:45

4aUW4. The acoustic energy conversion efficiency of a single raindrop. Dajing Shang, Qi Li, Shu Liu, and Fangzhou Deng (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Nangang District Nantong St. No. 145 Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn)

The measurement of rainfall in the ocean is more difficult than in land, but the noise signals produced by rainfall can be used for measuring the rainfall in the ocean. In this paper, a single raindrop measurement system was set up to measure the noise of the raindrop, the bubble sound is classified by the radius of the raindrop which was measured by the splash method. The mechanism of the initial impact and the bubble sound are analyzed, and the kinetic energy threshold and acoustic energy conversion efficiency of the single raindrops is also investigated. The results show that the bubble sound is the main compared with initial impact sound, the initial impact sound will become larger with the radius of raindrop, the bubble acoustic energy of large and great raindrop is much larger than that of small one, but the tiny and medium raindrop have no bubble acoustic energy. The kinetic energy threshold is not a constant, but proportional to the raindrop size. The acoustic energy conversion efficiency is about 1.04×10^{-3}% for small raindrops, but 10^{-5}% for heavy rain and great raindrops.

9:00

4aUW5. Underwater ship noise pattern detection and identification. Julia Berdnikova (Thomas Johann Seebeck Dept. of Electronics, Tallinn Univ. of Technol., Ehitajate tee 5, Tallinn 19086, Estonia, julia.berdnikova@ttu.ee), Aleksander Klauson, Mirko Mustonen (Civil Eng. and Architecture, Tallinn Univ. of Technol., TALLINN, Estonia), and Mihkel Tommingas (Thomas Johann Seebeck Dept. of Electronics, Tallinn Univ. of Technol., Tallinn, Estonia)

The anthropogenic underwater noise event could be identified by spectral analysis and acoustic pattern pre-classification of previously measured noise sources. This paper discusses real-time applicable methods for separation of purely natural noise recordings when ships are absent from the data polluted by the ship noise. The natural noise periods are used for the statistical modeling of underwater channel environment and relative levels of marine ambient noise (Wenz curves). The anthropogenic noise periods allow us to identify the noise sources. Moreover, most of the ship identification methods require spectrum component processing and pre-determined ship classification, which could be based on this automatic separation results. The correlational and multiple criteria detection methods compared with cyclostationary feature detection and widely used energy detection. False alarm and misclassification minimization could be achieved by multiple sensor data fusion: weather and Automatic Identification System (AIS). The measured data were collected near the seafloor with an autonomous long-term passive acoustic recorder and were combined with ship passage information from the AIS. Three years of measurements include multiple individual ship observations with different speeds and directions. Analysis of the experimental results emphasizes the importance of pre-classification, especially in multiple target cases.

9:15

4aUW6. Ship source strength estimation in shallow water. Aleksander Klauson and Mirko Mustonen (Civil Eng. and Architecture, Tallinn Univ. of Technol., Ehitajate tee 5, Tallinn 19086, Estonia, aleksander.klauson@ttu.ee)

Continuous underwater noise from the commercial ship traffic is an important pressure on the marine environment. The environmental assessment of this pressure often includes both measurements and modeling. In order to model ship noise based on AIS traffic data, it is essential to provide a better Source Level (SL) input values for the individual ships. The SL can be estimated using underwater noise recordings along with the identified position of the ships. A good estimate of the SL in the deep water can be obtained by application of the spherical spread model. In the shallow water such an approach is inapplicable because of multiple interactions of the sound waves with the sea surface and the bottom and therefore sound propagation modeling should be applied. If the modeling is performed for some particular frequency bands, estimation of the transmission loss needs even more calculation effort. Simplified approach for more efficient calculation of the losses is proposed. The calculated results are compared with the measurements in different geographical positions and sea conditions. The topic of the ship source directionality is addressed. Repeatability of the results is checked for the different passages of the same ship during the year.

9:30

4aUW7. Resonant subwavelength acoustic panels with air inclusions for abatement of noise from underwater machinery and remotely operated vehicles. Colby W. Cushing, Kevin M. Lee, Andrew R. McNeese, Michael R. Haberman, and Preston S. Wilson (Appl. Res. Labs and Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu)

Deeply subwavelength thickness metamaterial panels consisting of encapsulated cylindrical air cavities cut from elastic plates were recently shown to provide broadband, low-frequency underwater noise suppression for frequencies above the resonance frequency of the cavity [*J. Acoust. Soc. Am.* **138**, EL245–EL257 (2015)]. The present work extends previous findings to investigate noise suppression applications associated with underwater machinery and remotely operated vehicles (ROVs). Air inclusions in the panels were sized for maximum attenuation of the frequency spectrum of the ROV and radiated pressure amplitude reduction as a function of frequency and void fraction was determined. To do this, panels were initially secured around an omnidirectional projector in a single discontinuous layer and the panel surface area was varied to quantify the effect of panel air volume fraction on the radiated noise level. Panels were then sized and arranged around an ROV and the measurements were repeated. Results will be presented and compared to an effective medium model. The system has proven to be a simple way to provide significant noise reduction for machinery and ROV-related noise which is of interest for shallow water applications. [Work supported by the US Navy Office of Naval Research.]

9:45–10:00 Break

10:00

4aUW8. Passive, broadband suppression of radiation of low-frequency sound. Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Plane gas-liquid interface becomes anomalously transparent for low-frequency sound, and essentially all energy radiated by a compact sound source in fluid is channeled into the gas when the source is within a fraction of wavelength from the interface [O.A. Godin, Phys. Rev. Lett. **97**, 164301 (2006)]. This is in contrast to high-frequency behavior, where the interface approximates a pressure-release boundary, and only a very small fraction of energy escapes the liquid. This paper investigates whether a similar effect of suppression of unwanted low-frequency sound radiation by underwater sources can be achieved in a wide frequency band by using compact, acoustically soft objects. Low-frequency asymptotics of acoustic Green's functions in the presence of spherical and cylindrical scatterers are used to quantify radiation suppression by simple shapes. Dependence of the radiation suppression efficiency in a homogeneous fluid and in underwater waveguides on mechanical properties of the soft objects is discussed. Feasibility of passive suppression of underwater sound radiation with and without employing acoustic metamaterials is addressed. [Work supported by ONR.]

10:15

4aUW9. In-water impact pile driving sound source spectra comparison and the quest for a "Generic Spectrum". Shane Guan (Office of Protected Resources, National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Melanie Austin (JASCO Appl. Sci. (USA) Inc., Anchorage, AK), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Noise generated from in-water impact pile driving for marine and coastal construction can impact marine mammals. Physiological effects to marine mammals include hearing impairments such as temporary and/or permanent threshold shifts. Because marine mammals' auditory response to sound levels is frequency-dependent, environmental impact analysis from noise exposure needs to consider noise spectrum in addition to broadband received level. In this study, spectral characteristics of measured noise levels from impact driving of 30-inch steel piles at four locations in Alaska and Washington were compared. Results show for all impact pile driving, most acoustic energies were between 100 and 1,000 Hz. Spectral levels below 100 Hz varied among the locations. Above 1,000 Hz, spectral levels decayed at a rate of 15 dB/decade for the three Alaska sites, but at 30 dB/decade in Washington. Despite these general characteristics, however, there does not seem to be a "generic spectrum" for the 30-inch steel pile impact driving. The variations among the spectral characteristics are likely due to differences in substrate, hammer energy, and bathymetry in different locations.

10:30

4aUW10. High-resolution spectral-element computation of underwater noises due to offshore piling. Arvin Manalaysay, Chanseok Jeong (Civil Eng., The Catholic Univ. of America, 20 Michigan Ave., N.E., Washington, DC 20064, manalaysay@cua.edu), and Hom Nath Gharti (Dept. of GeoSci., Princeton Univ., Princeton, NJ)

Offshore piling has been effective in building foundations of offshore structures, such as wind turbines, bridges, and oil rigs. Despite such merits, underwater noise due to offshore piling is considered to be its critical setback. Pile driving creates a high-level underwater sound that harms marine ecosystems. There have been studies that successfully predicted these noises by using numerical methods, for example, Finite Element Method (FEM). However, there has been no FEM study that considers anti-symmetric irregular domains and complex bathymetry to compute underwater noises in the high-frequency range (>1000 Hz) due to expensive computational costs of FEM. To bridge the gap, this work attempts to explore a novel, powerful simulation tool to efficiently obtain offshore piling noises in complex settings and in the high-frequency range. We adopted and modified an open-source large-scale parallel Spectral Element Method (SEM) wave simulator, SPECFEM3D. SEM is known to be much more efficient than FEM for wave propagation analysis problem of a very large number of elements and time steps without compromising accuracy. Our computational method can be used for prediction of offshore piling underwater noise and investigating novel piling methods, such as optimized shapes of piles or air bubbles curtains to mitigate underwater noise.

10:45

4aUW11. Sound radiation from a finite cylindrical shell with an irregular-shaped acoustic enclosure. Desen Yang, Rui Zhang, Shengguo Shi, and Tengjiao He (Underwater Acoust. Eng., Harbin Eng. Univ., NanTong Str NanGang Dist, Harbin 150001, China, 474996229@qq.com)

In practical situations, large machinery equipments are usually located in underwater vessels and change the regular-shaped cavity into an irregular one. Due to the existence of the machinery equipments, the sound transmission and radiation are difficult to express analytically. This paper models and analyses the noise radiation of a cylindrical shell excited by an internal acoustic source. The cylindrical shell contains a machinery equipment modelled as a rectangular object attached to shell with a spring-mass system. The acoustic field of cavity is computed by an integro-modal approach. Effects of object size on the coupling between acoustic modes and structural modes are investigated. Meanwhile, the relationship between volume of the object and sound radiation is studied. Numerical results show that the existence of the object gives rise to a more effective coupling between the structure and the enclosure compared with a regular-shaped cavity.

THURSDAY AFTERNOON, 7 DECEMBER 2017 — STUDIO 9, 1:00 P.M. TO 5:35 P.M.

Session 4pAA

Architectural Acoustics: Back to the Future: A Look at Multipurpose Spaces, How They've Changed, and What's Next

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Jennifer Nelson Smid, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. A brief history of the multi-use Theater form. Robin S. Glosemeyer Petrone (Threshold Acoust.com, 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

Multi-use theatres provide communities with a venue to accommodate almost any performance type that requires a stage. The advent of structural steel, mechanical ventilation, and amplifications systems are a few advances of the late nineteenth and early twentieth centuries that have given shape to the multi-purpose theater; how have they shaped the form we see today?

1:25

4pAA2. Active acoustics in multi-purpose venues: A ten year retrospective. Steve Ellison and Melody Parker (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

Multi-purpose venues better serve diverse programs by tailoring their acoustics for each use condition. Active acoustic systems use electronics to provide wide-ranging acoustic adjustability with the press of a button. Over the past decade, the Constellation acoustic system by Meyer Sound has been deployed in over one hundred venues, supporting the performing arts, worship, education, and corporate sectors. This paper presents a survey of such venues and discusses lessons learned.

1:45

4pAA3. The interactive multipurpose performing arts hall. Hyun Paek, Gary W. Siebein, Jennifer R. Miller, Marylin Roa, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com)

Each multipurpose performing arts hall has a unique acoustic signature. The success of a modern multipurpose hall relies on how the disciplines of architecture, interior design, stage lighting and rigging systems, and acoustical design are coordinated to very small tolerances. The shaping and finish materials are integrated both visually and with precision to create the sound field of the room to accommodate a variety of venues. Two case studies are presented. The first is a 1000 seat multipurpose performance hall in central Florida which accommodates the use of variable acoustics, a specially designed orchestral shell, stage lift, and all visible interior spaces sculpted to provide richness of both amplified and natural acoustics. The second is a performance hall for a magnet high school for the performing arts in western Georgia that has undulating waves of aesthetically integrated panels on the walls that provides a stunning multipurpose space even with economic constraints. The case studies present different approaches to acoustical design of multipurpose theaters. The first multipurpose theater accommodates the needs of varying visiting performers while the second case study focuses on the presentation of students of performing arts that needed an economical but aesthetic center piece of a magnet school. This interactive process between the design team ultimately results in the beauty of the interactive participation of the audience and the performers also.

2:05

4pAA4. Changing an existing performance space from a single purpose use to a multipurpose space or creating a multipurpose space from an existing performance space not acoustically suitable for any performances. Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, bob@rccoffeen.net)

Occasionally, a performance space is created for a particular use such as drama and later it realized by the owner and the users that the space should be acoustically suitable for multipurpose use. Also performance spaces are encountered that do not suitably serve performances of any type from an acoustical viewpoint but that must properly serve performances of various types. Examples of both situations are discussed and acoustical data for both situations will be presented.

2:25

4pAA5. Effects of variable acoustic elements on the spatial sound field in multipurpose venues. Michelle C. Vigeant, Matthew T. Neal, and David A. Dick (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu)

The design of multipurpose venues typically includes variable acoustic elements which adjust the acoustics of the space to support the intended use. The most common approach to vary the acoustics of a venue is to use variable absorption, often in the form of heavy drapery and/or acoustic felt banners. These elements are often characterized by changes in reverberation time, but their position also impacts the spatial distribution of room energy. In order to study the effects of variable acoustic elements on the three-dimensional sound field at specific audience locations, spatial impulse response measurements were taken in a number of venues using a 32-element spherical microphone array. The venues ranged from a small recital hall of 400 seats to a typical multipurpose hall with 1300 seats to a 2500-seat concert hall with variable acoustics. Beamforming techniques were used to analyze the effects of the variable acoustic elements on the spatial and temporal distribution of sound energy at several receiver locations in each venue. The sound fields in these venues will be compared to those measured in dedicated concert halls, which have been measured for recent work on the topic of listener envelopment. [Work supported by NSF Award 1302741.]

2:45

4pAA6. Multi-purpose performance spaces as vehicles to enhance the acoustical communities of cities and towns. Gary W. Siebein, Marylin Roa, Jennifer R. Miller, and Matthew Vetterick (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

Soundscape theory can offer insights into methods that can transform the design of multi-purpose performance venues so that they can come closer to fulfilling their potential as a vibrant facility at the artistic and cultural core of a community every day of the year all day long. This is a design theory guided by technical innovation and philosophical insight that can bring together diverse groups of users and performers; experimental building, acoustical and theatrical systems; and sophisticated, integrative design methods. Case studies of several architectural design studio projects that ask questions about the nature of performance in the city will be used to describe an approach to integrated architectural and acoustical design based in soundscape theory that brings forth new approaches to using sound to shape space rather than designing spaces and inserting acoustical envelopes within the spaces based on transformative mapping of interior and exterior space created by sound and the ways that it is enhanced when it interacts with buildings.

3:05

4pAA7. Trends in performing arts center design. Jack P. Hagler (Theatre Planners, Schuler Shook, 325 N Saint Paul St., Ste. 3250, Dallas, TX 75201, jhagler@schulershook.com)

The world of performing arts continues to evolve. Audiences have changed. With every show, contemporary live entertainment delivers an updated, but still unique, experience. Operas of the 18th century are the Broadway shows of the 21st century. Orchestras of the 1700s are the Thievery Corporation of today. Lillie Langtry is now Lady Gaga. Performing arts centers must embrace the concept of popular entertainment inside and outside of the audience chamber to keep audiences excited, engaged, and attending. Contemporary entertainment attracts a contemporary audience—a technology-driven generation with a new set of expectations when it comes to a night at the theatre. Here are trends we've observed in our practice. Convertible audience rooms and floors for more versatility and flexibility; PAC reputation, space utilization, and the decline of attendance for classical symphony orchestra programming; audience immersion; creating a more social experience in the audience chamber and beyond; making a night (and day) of it with better food and beverage offerings; safety, security, and the new norm; animation of architecture. This is a discussion of the shift paradigm of performing arts centers from the classic arts to the contemporary arts and how the acoustician might respond to these contemporary trends.

3:25–3:40 Break

3:40

4pAA8. Case study: Eccles Center for the Performing Arts. Ronald Freiheit and Matthew S. Hildebrand (Wenger Corp., 555 Park Dr., Owatonna, MN 55060, ron.freiheit@wengercorp.com)

Cinemas and movie theatres have specific room acoustic requirements that are well understood in the industry. Recent technologies such as Dolby Atmos and DTS:X provide an exciting spatial audio experience using a high number of audio tracks dynamically rendered to loudspeakers located in both the overhead and lateral planes. But what happens when the same venue is utilized for both motion picture screenings and acoustic performances? In this paper, active acoustics processing is integrated into existing cinematic surround sound architecture to provide a unique solution that promises flexibility rather than compromise. Design objectives, limitations, and performance of the integrated systems are reviewed.

4:00

4pAA9. Jazz: Multipurpose art form meets multipurpose space. Evan Delli Paoli (Holzman Moss Bottino Architecture, 90 BRd. St., Ste. 1803, New York, NY 10004, edellipaoli@holzmanmossbottino.com)

The contemporary multipurpose performance space is not one that can exist without its evolution from Ancient Greek Amphitheaters to Elizabethan theaters to roaring twenties Movie Palaces. Its accelerating trajectory and relevance are directly correlated to the evolution of the art forms taking place and unfolding from within them. One such art form is Jazz. This session will explore episodic moments throughout the history of the Jazz art form that draw parallels to pivotal moments in the evolution of multipurpose performance spaces as we know them. The session is organized around five specific case studies: (1) Dixieland and The Rise of Armstrong; (2) Ellington, The Jazz Orchestra, and The Music Halls; (3) Miles: Birth of the Cool, Bitches Brew and Jazz Reinvention; (4) Gillespie and the melting-pot of Afro-Cuban Jazz; and (5) Jazz Fusion and found space. In each case study, a moment in Jazz history will be used as an analogy to introduce the architectural and acoustic concepts specific to a particular typology of multipurpose space—accompanied by a specific built building project example of how these concepts physically manifest themselves.

4:20

4pAA10. Hybridization of performance venues. Wendy Pautz and Julie Adams (Architecture, LMN Architects, 801 Second Ave., Seattle, WA 98104, jadams@lmnarchitects.com)

It should come as no surprise that the future of the performance space is wedded to changing demographics and increasingly pluralistic artist expression. New models are needed to survive the rapid societal change—socially, culturally, and economically. While multipurpose functionality has long since become the norm for both civic and community arts venues, we are now seeing the advent of hybrid civic facilities—venues that serve other forms of public gathering to complement arts programming. This proposition entails rethinking performance venue typologies at a fundamental level to ensure a sustainable future. From the multiple perspectives of a diverse panel of industry experts, this session will explore the underlying factors that shape current and future venue trends, including capacity to serve a wide range of artistic content as well as other civic and community programming. The following project case studies will be examined as reference points to the discussion: *The Tobin Center for the Performing Arts, San Antonio Federal Way Performing Arts and Event Center, Federal Way, Washington Octave 9 (experimental performance venue under development for the Seattle Symphony Orchestra)* Audience engagement and spirited dialogue around this topic is encouraged and expected!

4:40

4pAA11. A trio of multi-purpose halls, 1995–2016: Aronoff, Overture, and Hancher. Joseph W. Myers (Kirkegaard Assoc., 801 W. Adams St. 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

The Aronoff Center in Cincinnati OH (1995) includes a 2700-seat multi-purpose hall used as a roadhouse, with a particular focus on touring Broadway shows. The Overture Center in Madison, WI (2005), has a 2250-seat multi-purpose hall that is the home to a professional symphony orchestra and opera company, as well as hosting touring Broadway shows. Hancher Auditorium in Iowa City IA (2016) has an 1,800-seat multi-purpose hall that acts as a roadhouse for a broad variety of performance types, as a presenter for dance, and as the largest performance space for the University of Iowa's arts programs. All three performing arts centers had Pelli Clarke Pelli as design architect, Theatre Projects Consultants as theater consultant, and Kirkegaard Associates as acoustics consultant. This presentation examines what these halls have in common, how they are different, and what drove these differences. It looks specifically at the degree to which the differences are a design response to the particular program of each room and the degree to which they reflect changes over time in design approach or design requirements.

5:00

4pAA12. Multi-purpose halls… Wave of a new future or dead-end? Paul H. Scarbrough and Christopher Blair (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, pscarbrough@akustiks.com)

Multi-purpose halls, that is, performance spaces that can adapt their acoustical and technical features to serve many different performance types are a uniquely North American invention. Europe, where the modern-day symphony orchestra, opera company, ballet company, and drama troupe all originated, favors halls that are purpose-built to serve each art form. In the United States and Canada, even in cities that could afford to field multiple ensembles or companies, building dedicated facilities for each art form was often a bridge too far. This gave rise to the multi-purpose hall and with it the old saw that multi-purpose meant no-purpose. The success of new venues in Charleston, San Antonio, León, México, and elsewhere demonstrates that multi-purpose halls are cost-effective solutions and that they need not be the compromises they once were. The authors will explore the development of multi-purpose halls starting with

Louis Sullivan's Auditorium Theater in Chicago, arguably the forerunner of the contemporary multi-purpose hall, through to the current day. The authors will also explore how the evolving creative impulses of artists and directors is challenging designers to go even further, and incorporate degrees of flexibility into these spaces that would have been unimaginable just two decades ago.

Contributed Paper

5:20

4pAA13. Sixteen years later—Skirball Center for the performing arts at NYU completes the installation of Electronic Architecture. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

The Skirball Center for the Arts at New York University was designed to be a multi-purpose venue with Electronic Architecture providing variable acoustics. Although the infrastructure (wiring, loudspeakers) was integrated during construction in 2000, the system was not completed and remained idle since that time. In 2016, New York University decided to complete the system installation. We will discuss the nature of the venue and its programming, the reasons that the system remained dormant for so long, and the changes that sparked interest in completing the system.

THURSDAY AFTERNOON, 7 DECEMBER 2017 SALON F/G/H, 1:00 P.M. TO 3:05 P.M.

Session 4pAB

Animal Bioacoustics: Neurophysiology of Echolocation

Dorian S. Houser, Chair
National Marine Mammal Foundation, 2240 Shelter Island Drive, San Diego, CA 92106

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. Biosonar gain control in odontocetes: Evoked-potential studies. Paul E. Nachtigall and Alexander Supin (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, nachtiga@hawaii.edu)

The biosonar of odontocetes processes echo signals with a wide range of echo levels. Several mechanisms that serve to compensate for the echo-level variation (biosonar gain control) have been revealed: (1) the adjustment of the emitted sonar pulse levels (the longer the distance to the target, the higher the level of the emitted pulse), (2) the release from forward masking of the echo by the preceding self-heard emitted pulse (the longer the distance to the target, the more release of the echo-response from masking), and (3) an active control of hearing sensitivity (the lower the echo level, the higher the sensitivity). Investigations using the auditory evoked potential technique have demonstrated that these processes manifest different aspects of the functioning of a common gain-control system. The control of the emitted sonar pulse level does not change the echo-to-emitted pulse ratio but makes forward masking able to function at various target distances. The active control of hearing sensitivity also adjusts forward masking to function at various target distances. These combined processes are capable of producing an effective reduction in the variation of the level of the response to the echo when the target strength and target distance vary within a wide range

1:25

4pAB2. Echo jitter delay discrimination in bottlenose dolphins (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), James J. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA), Dorian S. Houser, and Ryan Jones (National Marine Mammal Foundation, San Diego, CA)

Experiments with bats show echo delay discrimination capabilities on sub-microsecond scales, which is an acuity that is much higher than that predicted by the auditory nervous system's ability to directly encode high-frequency phase information (i.e., through neural phase locking). To provide a cross-species comparison, echoic discrimination experiments were conducted with two bottlenose dolphins. Dolphins were trained to echolocate a "phantom" target and report a change from a target with a static delay of ~12 ms to one with alternating delay, or "jitter," imposed on the static delay. Performance was measured as a function of jitter delay. Both dolphins were able to reliably detect jitter down to ± 1 μs, although lower amounts of jitter were not detected. When the jittered echo was phase shifted by 180° relative to the static echo, performance was near 100% at all jitter delays, including ± 0 μs. Both dolphins utilized intermittent

patterns of click emissions, which were unusual for the relatively short ranges employed. Bottlenose dolphins do not appear to possess the sub-microsecond delay acuity capabilities observed in some bats. However, the current results suggest that dolphins can encode information resembling echo phase information in determining target range. [Funding from ONR.]

1:45

4pAB3. Narrowband auditory brainstem responses to "self-heard" and external clicks in the bottlenose dolphin. Dorian S. Houser, Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Carolyn E. Schlundt (Government IT Services, San Diego, CA)

Knowledge of the emitted click or chirp in echolocating dolphins and bats is believed to be critical to using echo delays to reveal target range and shape. In dolphins, the neural representation of the "self-heard" click should not equate to that of a click measured in the far-field due to sound propagation path differences and temporal dispersion within the cochlea. To investigate these differences, auditory brainstem responses (ABRs) to self-heard clicks masked with noise bursts having various high-pass cutoff frequencies were measured in two dolphins. A passive listening experiment was also conducted in which similarly masked external, spectrally "pink" clicks were used as stimuli. In both experiments, narrowband ABRs were derived using a high-pass subtractive noise technique. Latencies of the ABR to the external click demonstrated frequency-dependent latency shifts. Latencies of the ABR to the self-heard click were delayed relative to those of the passive listening experiment and similar across frequencies from ~28 to 113 kHz, suggesting that neural responses to the self-heard click are synchronous within the bandwidth of echolocation. Longer ABR latencies are potentially due to spectral differences between external and self-heard clicks, click-induced forward masking, and possibly, neural inhibition associated with click production.

2:05

4pAB4. Selectivity of bat midbrain neurons to stimulus elements embedded in natural echolocation sequences. Silvio Macias, Jinhong Luo (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

We investigated response selectivity of single neurons in the inferior colliculus (IC) of the big brown bat, *Eptesicus fuscus*, to acoustic elements embedded in the natural temporal patterning of dynamic frequency modulated (FM) echolocation sequences. Acoustic stimuli consisted of species-specific sonar sequences, with natural dynamic spectro-temporal features, containing both bat-emitted FM pulses and echoes from a moving target. Roughly 50% of sampled IC neurons showed responses to a restricted subset of stimulus elements in the natural sonar sequence and selectivity to the time delay between pulses and echoes, the bat's cue for target distance. Importantly, response selectivity emerged only when stimulus elements were embedded in natural sequences and was absent when the same neurons were stimulated with isolated pulse-echo pairs presented randomly at 300 ms stimulus intervals. These data provide compelling evidence that natural and dynamic sound sequences are necessary to evoke selectivity to pulse-echo delay in midbrain neurons of an FM bat. Our findings suggest that the FM bat's fine temporal control of pulse interval during insect pursuit contributes to neuronal selectivity that can serve to sharpen acoustic imaging by sonar.

2:25

4pAB5. Dynamic representation of 3D auditory space in the midbrain of the free-flying echolocating bat. Ninad B. Kothari, Melville Wohlgemuth, and Cynthia F. Moss (Johns Hopkins Univ., 3400, N. Charles St., Ames Hall Rm. 124, Baltimore, MD 21218, ninadbkothari@jhu.edu)

Our natural world is three-dimensional. A fundamental requirement of spatial orientating behaviors in the natural environment is the representation of 3D sensory space. Despite the importance of 3D sensory coding of a natural scene to guide movement, most neurophysiological investigations of this problem have been limited to studies of restrained subjects, tested with 2D, artificial stimuli. Here we show for the first time that auditory neurons in the midbrain superior colliculus of the free-flying echolocating bat encode 3D egocentric sensory space, and that sonar-guided inspection of objects in the environment sharpens spatial tuning of single neurons. Combining wireless multichannel neural recordings from free-flying bats, synchronized with video and audio data, and an echo model that computes the flying animal's instantaneous, stimulus space, we demonstrate 3D echo-evoked receptive fields of single auditory midbrain neurons in animals orienting in a complex environment. We discovered that the bat's active sonar inspection of objects dramatically tightens range tuning of single neurons and shifts peak activity to represent closer distances. Our research demonstrates dynamic 3D space coding in a freely moving mammal engaged in a real-world navigation task.

2:45

4pAB6. Non-auditory, electrophysiological potentials preceding biosonar click production in bottlenose dolphins. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, Ryan Jones, Dorian S. Houser, Alyssa W. Accomando, and Sam H. Ridgway (National Marine Mammal Foundation, San Diego, CA)

The auditory brainstem response (ABR) to a dolphin's own emitted biosonar click may be measured by averaging epochs of the instantaneous electroencephalogram (EEG) that are time-locked to the emitted click. In this study, waves in the averaged EEG preceding the biosonar click-evoked ABR were measured using surface electrodes placed on the head in six configurations while dolphins performed an echolocation task. Simultaneously, clicks were measured using contact hydrophones on the melon and a hydrophone in the farfield. The results revealed an electrophysiological potential (the pre-auditory wave, PAW) preceding the production of each biosonar click. The largest PAW amplitudes occurred with the non-inverting electrode just posterior of the blowhole and right of the midline — the apparent side of biosonar click generation. Although the source of the PAW is unknown, the temporal and spatial properties rule out an auditory origin. The PAW may be a myogenic potential associated with click production; however, it is not known if muscles within the dolphin nasal system can be actuated at the rates reported for dolphin click production, or if sufficiently coordinated and fast motor endplates of nasal muscles exist to produce a PAW detectable with surface electrodes. [Work supported by ONR.]

Session 4pAO

Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics: Biological Effects on Seabed Geoacoustic Properties II

Kevin M. Lee, Cochair
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Kelly M. Dorgan, Cochair
Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528

Invited Papers

1:00

4pAO1. Impact of organic matter on the physical and dynamic properties of marine mud. Richard H. Bennett (SEAPROBE, Inc., 501 Pine St., Picayune, MS 39466, rhbenn_seaprobe@bellsouth.net), Matthew H. Hulbert (Res. Dynam., West Chester, PA), and Roger Meredith (Retired, Slidell, LA)

The physical and dynamic properties of marine mud are a function of the interaction of clay fabric, organic matter (OM), and seawater physico-chemistry, and, when present, free gas. OM is the focus of this presentation and is a determinant of several properties in mud deposits including (1) *free water* available versus total water content in clay fabric pore space as a result of (2) *OM seawater hydration,* (3) reduction of *permeability* by volumetric contribution of hydrated OM in pore space, (4) *OM density* reduction *by seawater hydration* versus dry OM density, (5) *dynamic behavior* by physico-chemical attachment of OM to clay particles in potential energy fields created by the clay fabric signatures (edge-to-face, edge-to-edge, and offset face-to-face) in seawater, (6) high percentages of OM (TOC >2%) result in a *high degree of compressibility* at considerably lower stress compared to mud deposits with <2%TOC. Measurement and characterization of OM, previously often neglected, is expected to provide new significant insights and understanding of marine mud physical and dynamic properties and contribute to more reliable research data for scientists and engineers. Handling and storage methods used for mud samples containing OM are critical for accurate results.

1:20

4pAO2. Biological effects and the Grain-Shearing model of wave propagation in unconsolidated sediments. Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

The Grain-Shearing (GS) model of wave propagation in unconsolidated sediments is embodied in two sets of dispersion relations, one for the compressional wave and the other for the shear wave. Besides frequency, these expressions for the wave speed and attenuation depend on the physical properties of the sediment, notably the bulk density and porosity. They also involve a material exponent, n, and grain shearing coefficients that are associated with the sliding of grains against one another, which is a non-linear, stick-slip type of process that is controlled by micro-asperities on the surface of the grains. Bioturbation is known to affect both the bulk density and porosity of sediments, while biogenic coatings of mineral grains modify the surface roughness, which alters the grain-shearing coefficients. The magnitude of the effects of bioturbation and biogenic coatings on the wave speeds and attenuations predicted by the GS theory will be discussed, based on available biological information in the literature. [Research supported by ONR.]

1:40

4pAO3. Microbiota and macrobiota alterations impact acoustic predictions of geotechnical sediment properties. Allen H. Reed (Marine Geosci. Div., Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529, allen.reed@nrlssc.navy.mil)

Computed tomography supplies a three-dimensional vision of sedimentary geometries that are fabricated by microscopic and macroscopic biota. The biota alter sediment structure and properties in such a way that the altered sediment structure impacts acoustic determinations of idealized seafloor properties, which are based upon grain size and associated, yet textbook, geoacoustic properties. The presence of these alterations affects propagation by scattering acoustic energy at density discontinuities and non-planar surfaces (aka ”seafloor roughness”) both of which have a likely impacts: dampening of signal strength and increases in reverberation. The result is that the information sought about the seafloor is obscured; sediment properties, composition, strength, and potential for deformation and mobility are ambiguous and poorly constrained from available acoustic information in areas where biological activity significantly alters the sediment structure. Here, the three-dimensional geometry of the microbiotic and macrobiotic impacts on sediments are presented in

images that display unique features, which are often discounted in sediment acoustic models, but which should be considered as important and highly relevant sources of uncertainty with respect to acoustic determinations of the seafloor geotechnical and geophysical properties.

2:00

4pAO4. Comparison between infauna abundance and seabed geoacoustic properties. Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Will M. Ballentine, Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

To investigate the effects of infauna on seabed geoacoustic properties, a field experiment was conducted during May 2017 in Petit Bois Pass, near the mouth of Mobile Bay, Alabama. *In situ* measurements of seabed geoacoustic properties were collected using hydrophones to generate and receive compressional waves from 5 kHz to 100 kHz and bimorph transducers to measure shear waves from 300 Hz to 1.5 kHz. The measurement system was deployed multiple times at three distinct sites within the pass, and acoustic measurements were conducted at depths ranging from 5 cm to 20 cm into the sediment. For each deployment of the acoustic system, diver cores were collected. A subset of the cores were sieved on site to collect infauna, and the remaining cores were taken back to the laboratory where they were sectioned and analyzed for porosity and grain size distribution. Comparison between compressional and shear wave speed and attenuation, the sediment geotechnical properties, and the distribution and abundance of infauna will be presented. [Work supported by ONR and ARL:UT IR&D.]

2:20–2:35 Break

2:35

4pAO5. Core and resonance logger (CARL) measurements of fine-grained sediments containing infauna. Gabriel R. Venegas, Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese, 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), and Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL)

Benthic infauna perform important ecological functions such as water filtration, the recycling of organic matter, and forage for larger predators. However, how infauna affect bulk acoustical properties of the sediment is less studied and warrants further investigation. During a field experiment conducted in Petit Bois Pass, AL, multiple cores of fine-grained sediments containing infauna were collected by divers and analyzed using a core and resonance logger (CARL). CARL is a core-logger capable of performing both conventional pitch-catch measurements radially along the core as a function of depth and lower frequency resonance measurements. Resonance measurements were performed by exciting acoustic modes within the core and sensing them externally along the core's length. By utilizing both pitch-catch transducers as receivers, symmetric and asymmetric modes can be identified, and sound speed can be inferred as a function of frequency. The frequency ranges for the pitch-catch and resonance measurements were 100 kHz to 400 kHz and 15 kHz to 30 kHz, respectfully. Subsequently, the cores were sectioned and were either sieved for infauna or analyzed for porosity and grain size distribution. The measured sound speeds will be compared with biological and geological properties of each individual core. [Work supported by ONR.]

2:55

4pAO6. Effects of marine infauna on the acoustic properties of sediment. Will M. Ballentine, Kelly M. Dorgan (Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, wballentine@disl.org), Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese, Preston S. Wilson, and Gabriel R. Venegas (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Marine infauna alter the surrounding habitat in many ways. Compact mud burrows, tubes built from shell hash, and large subsurface galleries are a few examples of these alterations. Structural changes such as these can have varying effects on the geoacoustic properties of sediment. Here, we investigate how infauna may affect the sound speed and attenuation in sediments both in natural diverse communities of organisms, and in laboratory mesocosm experiments using controlled monocultures of species representing potentially important functional groups. Field studies were conducted at Petit Bois Pass off the coast of Dauphin Island, Alabama in May 2017 in which sediment cores were collected and brought back to the lab for acoustic measurements. These measurements can be directly compared with those taken *in situ* using a deployable field apparatus. For sediments with both natural and manipulated communities of infauna, sound speed and attenuation were measured at multiple depths and at high frequencies with wavelengths corresponding to the scales of expected impacts of individual organisms (100–400 kHz) to assess the effects of different infaunal functional groups.

3:15

4pAO7. Sediment characterization using normal-incidence echo sounding in a biologically active environment. Marcia J. Isakson, Michael Rukavina, and Johanna Owens (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

In collaboration with the Dauphin Island Sea Lab, normal-incidence acoustic measurements were taken in the Petit Bois Pass which is know to harbor benthic biology such as tube worms. The acoustic system used is easily deployed from a small craft and self-calibrating. The acoustic measurements were taken coincidentally with grab samples to characterize both the sediment type and the types of benthic biology. Data were also collected in an area void of benthic biology for comparison. These measurements were analyzed to assess the ability of normal-incidence measurements to determine sediment type in the presence of benthic biology and also to develop methods for characterizing benthic biology with acoustics. [Work sponsored by ONR, Ocean Acoustics.]

3:35

4pAO8. Low-frequency acoustic behavior of photosynthetically active seagrasses. Jay R. Johnson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Jean-Pierre Hermand (LISA Environ. HydroAcoust. Lab, Université libre de Bruxelles, Brussels, Brussels Capital, Belgium), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic remote sensing techniques are an important tool for mapping seagrass coverage. Three different photosynthesis-related processes can occur in seagrasses that affect the acoustic behavior. First, the air channels within the leaf pressurize with produced gas. Next, bubbles form on the leaf blades. Finally, some of these bubbles break off and enter the water column. A one-dimensional acoustic resonator technique was adapted to monitor the photosynthetic activity of two Mediterranean seagrasses, *Posidonia oceanica* and *Cymodocea nodosa*. Measurements of the low-frequency (1–8 kHz) effective sound speed of a mixture of seagrass leaf blades and artificial seawater were taken at regular intervals during periods of no direct light and exposure to photosynthetically active radiation. The acoustic response is compared to independent dissolved oxygen measurements and visual observation of bubble formation.

3:50

4pAO9. Variations in ultrasonic transmission behavior along seagrass leaf blades. Jay R. Johnson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Jean-Pierre Hermand (LISA Environ. HydroAcoust. Lab, Université libre de Bruxelles (ULB), Brussels, Brussels Capital, Belgium), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Seagrass is a complex multi-phase material, and an effective method for connecting acoustic propagation through seagrass meadows to internal or external characteristics of the seagrass would be beneficial for acoustic remote sensing applications. To investigate some of these connections, ultrasonic (1, 2.25, and 5 MHz) time-of-flight measurements through individual leaf blades of the endemic Mediterranean seagrasses *Posidonia oceanica* and *Cymodocea nodosa* are presented. Acoustic measurements were made at multiple points along the leaf blades and the sound speed and signal attenuation varied significantly within a single blade depending on measurement location. The measured acoustic variations are compared to external blade features such as discoloration, epiphyte coverage, and thickness. Microscopy images of blade cross-sections each taken at each acoustic measurement location are used to compare the void fraction to acoustic behavior. [Work supported by ONR, ONR Global.]

4:05

4pAO10. Acoustical characterization of a seagrass meadow in the Lower Laguna Madre. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese, Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Abdullah F. Rahman (School of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX), Justin T. Dubin, and Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This talk presents preliminary results from an experiment conducted in the Lower Laguna Madre, Texas to characterize the physical and acoustical properties in a meadow of *Thalassia testudinum.* Concurrent measurements were collected using (1) acoustic probes, (2) side-scan and parametric sonar, (3) broadband propagation, and (4) sediment cores. The acoustic probes provided localized, direct measurements of sound propagation in the seagrass canopy as well as geoacoustic properties (compressional and shear wave speed and attenuation) of the seagrass-bearing sediment. The side-scan and parametric sonars were used to survey for seagrass abundance and sub-bottom layering. Broadband signals produced by a combustive sound source were recorded at several ranges by hydrophones and geophones and were used to infer geoacoustic properties of the seagrass and underlying sediment for rapid environmental assessment. The sediment cores were analyzed in the laboratory using both low-frequency resonator measurements and high frequency travel-time measurements to estimate compressional wave speed, after which they were sectioned and measurements of sediment grain size, porosity, and biomass were obtained. The combination of these data sets provides a unique characterization of the geoacoustic properties of a seagrass meadow. [Work sponsored by ARL:UT IR&D and ONR.]

4:20–4:40 Panel Discussion

Session 4pBA

Biomedical Acoustics: Imaging II

Marie M. Muller, Chair
MAE, NCSU, 911 Oval Drive, Raleigh, NC 27695

Contributed Papers

1:00

4pBA1. Application of laser ultrasound technique to evaluate wave velocity in bovine meniscus. Yoshitaka Sakata (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, duq0356@mail4.doshisha.ac.jp), Shunki Mori, Mami Kawase, and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

In vivo elasticity evaluation of meniscus and cartilage is necessary to understand the condition of the joint. Ishihara *et al*. have used the photoacoustic method with an endoscope.[1] Another possible technique is the laser ultrasound. Making use of the thermoelastic effect, we can stimulate a short longitudinal wave by irradiating the pulsed laser beam to the material. This technique enables noncontact measurements. As an initial study, we have measured the longitudinal wave velocities in a bovine meniscus sample using a pulsed laser (COHERENT Helios 1064-5-50). The meniscus samples were obtained from the knee joint of a bovine femur. The sample size was $4.00\times6.50\times6.00$ mm^3. We could excite short longitudinal waves around 30 MHz and measure wave velocities propagating in three directions (bone axis, posterior-anterior, and medial-lateral). The excited ultrasonic wave propagated through the meniscus sample and was received by a piezoelectric sensor. The velocity range was about 2950 m/s–3450 m/s, showing the highest velocity in the anterior-posterior direction. Velocities also depended on the water content and showed small heterogeneity. [1] M.Ishihara *et al*., Jpn. J. Appl. Phys. **42**, 556–558 (2003).

1:15

4pBA2. Time domain analysis of ultrasonic backscatter signals from cancellous bone. Jordan P. Ankersen, Phoebe C. Sharp, Joseph A. McPherson, Ann Viano, and Brent Hoffmeister (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, ankjp-18@rhodes.edu)

Ultrasonic backscatter techniques are being developed to detect changes in bone caused by osteoporosis. Most techniques analyze backscatter signals in the frequency domain by measuring quantities related to the power spectrum. Investigate the utility of two backscatter parameters determined from a time domain analysis of backscatter signals: the normalized backscatter amplitude ratio (nBAR) and the backscatter amplitude decay constant (BADC). A 3.5 MHz transducer was used to acquire backscatter signals from 54 specimens of bone prepared from 14 human femurs. nBAR was determined from the log of the ratio of the root mean square amplitude of two different portions of a backscatter signal. BADC was determined by measuring the exponential decay in the amplitude of a backscatter signal. nBAR and BADC both demonstrated highly significant ($p < 0.001$) linear correlations with bone density. However, the correlation coefficients were slightly stronger for nBAR ($0.79 \leq R \leq 0.89$) than for BADC ($0.67 \leq R \leq 0.73$). Parameters based on a time domain analysis of backscatter signals from bone may be sensitive to changes in bone caused by osteoporosis. Of the two parameters tested, nBAR demonstrated the strongest correlations with bone density. [**Funding**: NIH/NIAMS R15AR066900.]

1:30

4pBA3. Correlation of ultrasonic backscatter difference parameters with bone density in clinical ultrasound images. Abel Diaz, Ann Viano, Joseph A. McPherson (Dept. of Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, diaaa-18@rhodes.edu), Brian S. Garra (Div. of Imaging, Diagnostics and Software Reliability, US Food & Drug Administration, Silver Spring, MD), and Brent Hoffmeister (Dept. of Phys., Rhodes College, Memphis, TN)

The ultrasonic backscatter difference technique analyzes the power difference (in dB) between two gated regions of a backscatter signal and is being used to detect changes in bone caused by osteoporosis. The current study investigates correlations with bone density for these parameters determined from clinical ultrasound data. Ultrasonic backscatter images and signals were acquired from the hip and two vertebral bodies of human subjects using a 2.5 MHz phased-array transducer. Three backscatter parameters-normalized mean, slope, and intercept of the backscatter difference- were determined from the power difference between two gated regions of the signal. All three parameters were analyzed for 25 different gate choices. X-ray bone mineral density data for each subject were acquired for the three anatomical regions. Significant linear correlations ($p<0.05$) were found for all three ultrasonic parameters for at least one choice of gate parameters. R values ranged from to 0.38 to 0.63. For the first time, backscatter difference parameters measured *in vivo* have been shown to correlate with bone density. The correlation may depend on the gated regions of the backscatter signal chosen for analysis. [Funding: NIH/NIAMS R15AR066900.]

1:45

4pBA4. Modeling of the acoustic radiation force in elastography. Fabrice Prieur (Dept. of Informatics, Univ. of Oslo, P.O box 1080, Blindern, Oslo 0316, Norway, fabrice@ifi.uio.no) and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Elastography is a non-invasive imaging technique that can assess *in vivo* tissue stiffness. Shear wave elastography imaging uses the acoustic radiation force (ARF) to produce shear waves. Numerical models rely on different formulations of the ARF. We present three existing formulations for the ARF: its full expression in the second-order approximation and two simplified formulations using a quasi-plane wave and an attenuated plane wave approximation. Analytical expressions for the ARF are derived for the special cases of a spherical concave source and for a quasi-Gaussian beam. They show discrepancies between the different formulations. For strongly divergent or highly focused beams the result from the second-order approximation differs from both simplified formulations. However the second-order and quasi-plane wave approximations create identical shear displacements. The k-Wave simulation package is used to compute the ARF and the ensuing transient displacement produced by an ultrasound probe. The second-order approximation and the quasi-plane wave approximation give different forces but identical displacements. The results using the plane wave

approximation significantly differ. It is concluded that for highly focused transducers only the second-order approximation accurately estimates the ARF. For estimating shear displacements the second-order or quasi-plane wave approximation are equivalent and preferable to the plane wave approximation.

2:00

4pBA5. Combined subharmonic and ultraharmonic intravascular ultrasound imaging. Himanshu Shekhar (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Jeffrey S. Rowan, and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Abnormal proliferation of the vasa vasorum has been implicated in the rupture of atherosclerotic plaques. Imaging the vasa vasorum could help clinicians identify rupture-prone plaques and guide the choice of therapy. We hypothesized that subharmonic and ultraharmonic modes can be combined to improve the performance of contrast-enhanced intravascular ultrasound (IVUS) imaging. To test this hypothesis, vessel phantoms perfused with phospholipid-shelled ultrasound contrast agent (Targestar-P®, Targeson, Inc., San Diego, CA) were visualized using either subharmonic, ultraharmonic, or combined subharmonic and ultraharmonic modes. Flow channels, as small as 0.8 mm and 0.5 mm in diameter, were imaged using commercial peripheral and coronary imaging catheters at 12 MHz and 30 MHz transmit frequencies, respectively. Subharmonic and ultraharmonic imaging modes attained contrast-to-tissue ratios (CTRs) of 18 $\pm$ 2 dB and 20 $\pm$ 2 dB at 12 MHz transmit frequency, and 9 $\pm$ 2 dB and 13 $\pm$ 1 dB at 30 MHz transmit frequency, respectively. Combining subharmonic and ultraharmonic modes enhanced the CTRs to 33 $\pm$ 3 dB and 25 $\pm$ 2 dB at 12 MHz and 30 MHz transmit frequencies, respectively. These preliminary findings support the continued investigation of combined subharmonic and ultraharmonic IVUS for vasa vasorum imaging.

2:15–2:30 Break

2:30

4pBA6. A novel coding scheme and its application to ultrafast ultrasound imaging. Yang Zhang, Yuexin Guo, and Wei-Ning Lee (Elec. and Electron. Eng., The Univ. of Hong Kong, Rm. 807, Chow Yei Ching Bldg., Pokfulam Rd., Hong Kong, Hong Kong 0000, Hong Kong, zhangy@eee.hku.hk)

Ultrafast ultrasound imaging using plane or diverging waves, not focused beams, has enabled quantitative assessment of biological tissue elasticity, kinematics, and hemodynamics beyond anatomical information in the past decade. However, its sonographic signal-to-noise ratio (SNR) and penetration depth are limited by insufficient energy delivery under safety limits. We hereby propose a novel coding scheme and apply it to ultrafast ultrasound imaging to increase the SNR without compromising the spatial resolution and frame rate. In our coded ultrafast ultrasound imaging scheme, each transmit is a long pulse containing N ($N = 2^k$, $k = 0, 1, 2,\ldots$) waves with short time intervals and polarity coefficients of $+1$ or -1, instead of the conventional short pulse with single wave. In reception, a linear decoding scheme comprised of addition, subtraction and delay operations is devised to recover N times higher intensity backscattered signals to gain SNR of $10\cdot\log10(N)$. Experimental results acquired by the Verasonics Vantage system from the calibration phantom and *in vivo* human back muscle show that the proposed method using two transmits of $N = 32$ waves achieves 13.0 $\pm$ 0.5 dB SNR improvement at 40 mm depth at 4000 frames/sec, leading to better image contrast and larger penetration depth.

2:45

4pBA7. Etiology of the color Doppler ultrasound twinkling artifact on *in situ* human kidney stones. Julianna C. Simon (Graduate Program in Acoust., Pennsylvania State Univ., University Park, PA and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Barbrina Dunmire, Bryan Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Jeffrey Thiel, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and James R. Holm (Ctr. for Hyperbaric Medicine, Virginia Mason Medical Ctr., Seattle, WA)

Hyperbaric pressures of 0.3–10 MPa (absolute) have been shown to reduce the magnitude of the color Doppler ultrasound twinkling artifact on *ex vivo* human kidney stones, supporting the hypothesis that surface crevice microbubbles cause twinkling. For the first time, we investigate the etiology of twinkling on *in situ* human kidney stones. Eight human subjects with kidney stones known to twinkle were imaged with a Philips/ATL P4-2 transducer and Verasonics ultrasound system for 45 minutes while inside a hyperbaric chamber. Subjects breathed ambient air while the pressure in the chamber cycled to a maximum pressure of 0.4 MPa, with a scheduled decompression stop at 0.16 MPa where subjects breathed pure oxygen. Preliminary results show no change in the twinkling amplitude at 0.4 MPa compared to the baseline measurements before pressurization; however, a statistically significant increase in the twinkling amplitude was observed when subjects breathed pure oxygen at 0.16 MPa ($p = 0.046$). The increase in the twinkling amplitude upon changing respiratory gas composition further supports the crevice microbubble hypothesis of twinkling. Higher pressures than explored in this study may be needed to reduce twinkling on *in situ* human kidney stones. [Work supported by the National Space Biomedical Research Institute through NASA NCC 9-58 and NIH grant DK043881.]

3:00

4pBA8. Recovery of the complete data set from ultrasound sequences with arbitrary transmit delays. Nick Bottenus (Biomedical Eng., Duke Univ., 1427 FCIEMAS, Box 90281, Durham, NC 27708, nbb5@duke.edu)

Ultrasound beamforming relies on particular models of propagation to convert samples of the backscattered field through time into spatial samples of an image. The most common model used is straight-line propagation of a focused wave, assuming a narrow steered and focused beam that propagates radially along a selected direction. This model describes the main propagating pulse but fails to capture the true spatial extent of the wave. The reconstructed image suffers from defocusing, reduced signal-to-noise ratio (SNR), and contrast loss. A method is proposed to model transmission as the superposition of individual transmit elements on the transducer array and to efficiently recover the "complete data set," individual element transmit and receive responses, from an arbitrary scan sequence. Standard diverging wave focusing is applied to the complete data set to produce an image independent of the applied transmit focusing. For a sequence with a fixed transmit focus, the result is a high-SNR, two-way focused image. The proposed method also improves upon other synthetic transmit aperture methods such as plane wave imaging in that it captures the full diffracting wave field rather than relying on approximate field models, maximizing SNR at depth. Phantom and *in vivo* images are used to demonstrate improvement.

3:15

4pBA9. Study of effects of ultrasound stimulation on platelet-like-particle mediated fibrin-network deformation. Aditya A. Joshi (Mech. & Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Eng. Bldg. 3, Raleigh, NC 27695, aajoshi4@ncsu.edu), Nandi Seema, Ashley Brown (UNC-NCSU Joint Dept. of Biomedical Eng., North Carolina State Univ., Raleigh, NC), and Marie M. Muller (Mech. & Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

An important function of platelets is their ability to bind to fibrin. Over time, platelets contract the fibrin network to induce clot collapse through a process known as clot retraction. Synthetic platelet-like particles (PLPs) created from ultralow-crosslinked (ULC) microgels by conjugating ULCs to a fibrin-specific antibody are capable of mimicking this ability of natural platelets to induce fibrin network collapse. However, the rate of clot retraction is low compared to natural platelets. We demonstrate that deformability of a tissue-mimicking phantom containing ULCs increased in presence of ultrasound stimulation. We observe that the deformability of the ULC was optimal for 1MHz stimulation and a 0.025 mg/mL microgel concentration. PLP-laden, ULC-laden, and microgel-free fibrin clots were created and exposed to ultrasound stimulation for a period of 72 hours. The second set of clots was also created and monitored for 72 hours without ultrasound exposure. Clots were imaged via CryoSEM at 24 and 72 hours after polymerization to determine the effects of the PLPs and ultrasound stimulation on fibrin-network collapse. CryoSEM analysis of clots demonstrated increased density and decreased porosity in the fibrin network structure in presence of ultrasound, indicating microscopic clot collapse. These results suggest the potential of combining PLPs and ultrasound stimulation to alter fibrin clot properties and could be used in the future to enhance wound healing outcomes.

3:30

4pBA10. Validation of a wide-angle parabolic model for shallow-focus ultrasound transducers. Joshua Soneson and Yunbo Liu (Appl. Mech., FDA, 10903 New Hampshire Ave., Silver Spring, MD 20993, joshua.soneson@fda.hhs.gov)

Recently, a novel numerical method was developed for a wide-angle parabolic equation which accommodates steep gradients and discontinuities in the pressure distribution of the source boundary condition (Soneson, IEEE Trans Ultrason., Ferroelectr., Freq. Control **64**, pp. 679–687, 2017). The method allows rapid computation of acoustic fields with improved diffraction modeling capability over the standard parabolic approximation of the Helmholtz equation with no additional computational overhead, and is free from oscillatory artifacts which previously precluded the use of wide-angle models with discontinuous sources. In this work the wide-angle model, using the expression of a converging spherical wave (with a sharp cutoff at the edge of the active area) as the source boundary condition, is validated against measurements obtained using a 200 μm membrane hydrophone. The ultrasound field was produced by a single-element transducer with f-number 0.7 at 1.5 MHz, which was selected to test the wide angle model's account of diffraction physics well outside the regime of the standard parabolic model. Close agreement between the measured and computed fields is shown along the central axis and in the focal plane. This analysis represents a critical component of the rigorous validation of the model, which will come with a forthcoming uncertainty analysis.

3:45

4pBA11. Mapping the electrical activity of the heart *in vivo* using ultrafast acoustoelectric imaging. Beatrice Berthon, Philippe Mateo, Nathalie Ialy-Radio, Mickael Tanter, Mathieu Pernot, and Jean Provost (Institut Langevin, ESPCI Paris, PSL Res. Univ., CNRS 7587, INSERM U979, 17 rue moreau, Paris 75012, France, jean.provost@espci.fr)

The heart is driven by an electrical activation that propagates in cells and triggers their concerted contraction. When this electrical activation pattern is altered, it can lead to debilitating diseases such as arrhythmia or heart failure. Yet, directly imaging the electrical activity of the heart remains difficult to achieve. We have recently introduced Ultrafast Acoustoelectric Imaging (UAI) that combines ultrasound plane wave emissions and the acoustoelectric effect, i.e., the modulation of electrical impedance by ultrasound waves, to map electrical current densities in real-time with 1-mm and 5-ms spatial and temporal resolutions, respectively. Herein, we present its application to direct mapping of the cardiac activation in isolated live rat hearts and in pig hearts *in vivo*. UAI was performed in isolated rat hearts (n = 4) and in open-chest pigs (n = 1) using a 5-MHz linear array ultrasound probe fitted to a 256 channel- Vantage ultrasound system (Verasonics, Inc., WA). Two electrodes positioned on the heart and fed to the Vantage system were used to detect electrical impedance variations caused by ultrasound plane waves, which were then backprojected to reconstruct images of electrical current densities. UAI images depicted the propagation of the electrical activation of the cardiac tissue as validated by the electrocardiogram. These first *in-vivo* results suggest that UAI may allow the emergence of new biomarkers for the diagnosis and pre-clinical study of cardiac activation diseases such as arrhythmias.

Session 4pED

Education in Acoustics, Physical Acoustics, Signal Processing in Acoustics, Engineering Acoustics, and Underwater Acoustics: Synthetic Aperture Sonar for Youngsters

Murray S. Korman, Cochair
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Gregory E. Coxson, Cochair
U.S. Naval Academy, Electrical and Computer Engineering Dept., Mail Stop 14 B, 105 Maryland Street, Annapolis, MD 21402

Chair's Introduction—4:00

Invited Paper

4:05

4pED1. Classroom demonstration of synthetic aperture and array concepts. Chad M. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu)

Although often time consuming to develop and present, classroom demonstrations are an effective tool used in physics education. This talk walks through a demonstration of the basic concepts of array processing and synthetic aperture sonar. An emphasis has been placed on limiting equipment and setup time required to implement this demonstration, while allowing the presenter break the discussion into several conceptual topic areas. Primary topics include pulse compression and temporal/range resolution, array aperture footprint and azimuthal/range resolution, influence of transmitter and receiver beampatterns, and the importance of temporal coherence and measurement platform motion. Hardware require for this demonstration can be made as simple as a speaker and a single electret microphone used with a common PC soundcard.

Contributed Papers

4:25

4pED2. The Coffee-Can radar at the naval academy. Gregory E. Coxson and Erich Keyes (Elec. and Comput. Eng. Dept., U.S. Naval Acad., Elec. and Comput. Eng. Dept. U.S. Naval Acad., M.S. 14 B, 105 Maryland St., Annapolis, MD 21402, coxson@usna.edu)

The Coffee-Can radar was designed by Professor Gregory Charvat as a simple low-cost radar system for small teams of students to build and test during an intersession course at MIT. The name derives from the use of coffee cans as transmit and receive antennas. Since the success of Charvat's course, the coffee-can radar has been used at a number of schools for hands-on experiences with radar. It is one of the small radar systems used in courses in Principles of Radar at the United States Naval Academy, where laboratory and project-based learning are necessary components of the educational experience. This paper will describe how the radar operates, and will explain the role this radar plays in the Naval Academy's Principles of Radar and EW course. Attendees will receive a demonstration of the radar's three modes: ranging, range-rate estimation, and synthetic aperture imaging; and will have the opportunity to operate the radar. Additional small radar systems used in the Principles of Radar course will be available as well.

4:45

4pED3. Demonstration of synthetic aperture sonar or radar using shallow water waves in a ripple tank with small cylindrical targets. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

In an ideal stripmap synthetic aperture radar SAR or synthetic aperture sonar SAS experiment, a collocated transmitter and receiver array, respectively, generate a linear frequency modulated LFM pulse chirp that (1) reflects off of point-like targets that are placed along a smooth surface in the x,y plane and (2) receives backscattered echoes off the targets. The array (located a height h from the surface) may be towed (SAS) or flown (SAR), or in some airborne acoustic applications placed on a vehicle or track. In the stop and hop approximation waves are transmitted and then echoes vs. times are received and recorded at a point location along the track. At discrete locations along the track a set of N echoes are recorded which are pulse compressed using a correlation process. Using a time correlation backprojection algorithm the echoes are used to predict an image of the targets, namely the two-dimension reflectance f(x,y). Using a ripple tank, point-like spherical pulses are generated at locations guided by a linear track. The echoes from several slender cylindrical rods are received by a linear capacitance-to-voltage converter needle like probe [McGoldrick, Rev. Sci. Inst. **42**, 359–361(1971)] and backprojection is performed to predict the target locations.

Session 4pNS

Noise, ASA Committee on Standards, and Structural Acoustics and Vibration: Wind Turbine Noise

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Kenneth Kaliski, Cochair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics, 13 Pine Tree Road, Wellesley, MA 02482

Chair's Introduction—1:00

Invited Papers

1:05

4pNS1. Measurements of underwater sound radiated from an offshore wind turbine. James H Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett Bay Campus URI, Narragansett, RI 02882, miller@uri.edu), Ying-Tsong Lin, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Kathleen J. Vigness-Raposa, Jennifer Giard (Marine Acoust., Inc., MIddletown, RI), and Tim Mason (Subacoustech Environ. Ltd., Southampton, United Kingdom)

Underwater acoustic and geophysical systems were deployed to monitor the operation of the Block Island Wind Farm (BIWF). The BIWF consists of five GE Haliade 150-6MW wind turbines each with 150 m diameter blades. The five wind turbines were laid out about 1 km apart in a southwest-to-northeast arc. Each turbine is equipped with a direct drive permanent magnet generator, with no gearbox coupled to the generator. These turbines are variable speed and have independent pitch control by blade. The equipment used to monitor the BIWF operation consisted of a towed array of eight hydrophones, two VLAs with four hydrophones each and a fixed sensor package for measuring particle velocity. This sensor package consists of a three-axis geophone on the seabed and a tetrahedral array of four hydrophones at 1 m from the bottom. Additionally, an acoustic vector sensor was deployed in mid-water. During operations in December 2016, an acoustic signal was detected by the tetrahedral array of hydrophones at a position 50 meters west of the southwestern-most turbine. The frequency of this signal was approximately 72 Hz and the rms sound pressure level was about 100 dB re 1 micropascal. [Work supported by Bureau of Ocean Energy Management (BOEM).]

1:25

4pNS2. Sound level impact studies for wind energy in NY State: A whole new world. Robert O'Neal (Epsilon Assoc., Inc., 3 Mill & Main Pl Ste 250, Maynard, MA 01754-256, roneal@epsilonassociates.com)

In 2012, New York State adopted regulations for any energy facility over 25 megawatts in electrical capacity ("Article 10"). Since that time at least 15 wind energy projects have entered permitting, with many of them deep into the required sound level studies prescribed by Article 10. The author is working on seven of these projects and will cover the key steps in the process ranging from the initial public involvement through submittal of an application. The sound studies for Article 10 have become one of the most thorough and complex in the industry. Key elements of the technical studies include multi-season existing condition measurement programs, sound level modeling, meteorological data analysis, literature reviews, regulatory comparisons, and evaluation of a multitude of additional sound-related criteria such as hearing impairment and speech interference. The depth of the sound studies appear to be expanding beyond that contemplated in the Article 10 regulations.

1:45

4pNS3. Public acceptance of wind energy: Impact of sound levels. Thomas R. Haac (RSG, Inc., 55 RailRd. Row, White River Junction, VT 05001, ryan.haac@rsginc.com), Matt Landis (RSG, Inc., Burlington, VT), Kenneth Kaliski (RSG, Inc., White River Junction, VT), Ben Hoen, Joesph Rand (Lawrence Berkley National Labs, San Francisco, CA), Jeremy Firestone (Univ. of Delaware, Newark, DE), Johannes Pohl, Gundula Huebner (Institut für Psychologie der Martin-Luther-Universität Halle-Wittenberg AG Gesundheits- und Umweltpsychologie, Halle (Saale), Germany), and Debi Elliot (Portland State Univ., Portland, OR)

Lawrence Berkeley National Laboratory led a survey of 1,729 individuals located within 8 km of utility scale wind turbines in the United States. The survey included respondents around both large and small wind projects throughout the country. The survey focused on social acceptance, procedural and distributional justice, landscape and sound perceptions and annoyance, and compensation. A total of 15 of the wind projects were modeled to estimate the sound levels at each respondent's home. Modeled metrics included background

sound levels, maximum one-hour sound levels, percentage of time the respondent is downwind of a turbine, and a long-term sound level estimate using the local wind project capacity factor. Statistical analyses were conducted to estimate the acoustical drivers (sound level and sound level difference above background) toward the propensity for annoyance, and how these were affected by non-acoustic factors (e.g., compensation, prior attitude toward the project, visibility, etc.).

2:05

4pNS4. Human postural sway results in response to audible and infrasound emissions from wind turbines. Peggy B. Nelson, Andrew Byrne, Matthew Waggenspack (Dept. of Speech-Language-Hearing Sci., Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Michael Sullivan (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Christopher Feist, and William Herb (St. Anthony Falls Lab., Univ. of Minnesota, Minneapolis, MN)

Fifty healthy adult subjects ages 21–73 years attended to audible and infrasound signals generated from a wind turbine, recorded at 600 m and re-created in the laboratory. Stimuli consisted of modulated and unmodulated audible sound at 50 dB SPL, as well as recorded and peak-enhanced infrasound at an overall level of approximately 85 dB SPL (peaks up to 95 dB SPL). Participants were tested for their postural stability, detection, and ratings of audible and infrasound emissions from turbines. They also completed pre- and post-testing surveys for symptoms of imbalance. All subjects were blind to the knowledge that the stimuli were recorded from wind turbines. No significant adverse effects from healthy adults have been noted to date. Some individuals reliably indicated detection of infrasound signals. A few participants indicated mild symptoms (a rating of 1 on a scale from 0 to 4) following the test. Testing of patients who report vertigo will begin soon. Detailed results and implications will be discussed.

2:25

4pNS5. Subjective assessment of wind turbine noise—The stereo approach. Steven E. Cooper and Chris Chan (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

The conduct of stereo measurements for both playback in high-quality headphones and in a hemi-anechoic room has been undertaken for a number of wind farms and other low-frequency noise sources as an expansion on the material previously presented at the Boston ASA meeting. The results of the additional monitoring, evaluation of modulation index, and subjective analysis of this procedure is discussed and identifies the benefits of monitoring noise complaints and assessments of wind farm noise in stereo.

2:45

4pNS6. Measurements of infrasound blade pass frequencies in the far field. Andy Metelka (Sound and Vib. Solutions Canada, Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Infrasonic BPFs from wind turbines do appear as far as 120 km away from the nearest wind turbine at relatively low pressure levels. Previous papers clearly identify BPFs inside homes showing that they are rotational components of wind turbines. Both seismic ground vibration simultaneous to infrasound pressure are measured in the extreme far field in order to understand transfer paths. Air borne measurements clearly dominate in the far field also indicating audible amplitude modulation has no contribution to BPFs at these distances.

3:05–3:20 Break

3:20

4pNS7. Infrasound blade pass frequency transmissibility measurements inside homes near wind turbines. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Penetration of wind turbine blade pass frequencies are compared at different homes and various rooms show how levels change with wind direction and wind speed. Rooms facing turbines exhibited higher penetration compared to basement rooms, indicating air borne transfer paths vs. ground borne vibration. Construction characteristics of different homes are also compared using transmissibility calculations. Older homes exhibited higher transmissibility while newer airtight homes, under certain conditions, had lower transmissibility.

3:40

4pNS8. Acoustic compliance with permit conditions—What does it mean? Steven E. Cooper (The Acoust. Group, 22 fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au) and les huson (L Huson & Assoc., Woodend, VIC, Australia)

The use of a dBA L90/L95 limit for operational wind farms, that rely upon a regression analysis of wind and noise that is averaged over time, does not relate to complaints of disturbance. The relevance of the wind farm noise contribution versus the true L90 background level versus wind speed, the percentage of time above the nominal permit condition, and the impact of special audible characteristics is discussed.

4:00

4pNS9. Preliminary estimates of acoustic intensity vorticity associated with a turbine blade rate. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Peter H. Dahl (Mech. Eng. Dept. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

We report on measurements recently made at the Wild Horse wind farm, near Ellensburg, WA, as part of short experiment to evaluate a device sensitive to acoustic particle velocity. Measurements were made at range of approximately 60 m, directly in front of (facing) a turbine, at a height 1.5 m above ground using a sound level meter (SLM) and vector sensor positioned within a few meters of each other. The SLM recorded a steady A-weighted SPL of 58 dB and C-weighted SPL between 70 dB and 84 dB, noting some low-frequency variability due to wind noise. The vector sensor measurement consists of 4 coherently-recorded signals, one from an omnidirectional microphone and three from a tri-axial accelerometer embedded in a lightweight 10-cm diameter sphere, which helps immunize this acoustic particle motion measurement from wind noise. Here we focus on combining velocity and pressure measurements to form acoustic vector intensity. Real (active) and imaginary (reactive) components of this field display temporal properties corresponding to the position of the 3-turbine blades. Time-varying vorticity in the intensity vector demonstrate existence of a signature in wind-turbine noise related to the blade-passing rate (1 Hz) that is not registered with conventional sound level measurements.

4:15

4pNS10. Wind turbines—A Cape Bridgewater residents' experience. Melissa Ware (PO Box 5355, Geelong, Victoria 3215, Australia, wmlyss@yahoo.com.au) and Steven E. Cooper (The Acoust. Group, Lilyfield, NSW, Australia)

In 2008, 29 2 MW wind turbines were built on neighboring properties, 850 m from our solid limestone home. Having a bilateral senso-neuro hearing loss may result in a different experience. The experiences of living near a wind farm, the consequences, and involvement in the Cape Bridgewater Study are discussed.

4:30

4pNS11. Revisiting the South Australian Environment Protection Authority 2013 Waterloo study using the Schomer principle. Mary L. Morris (PO Box 188, Eudunda, SA 5374, Australia, morrisdg@outlook.com) and Steven E. Cooper (The Acoust. Group, Lilyfield, NSW, Australia)

In 2013, the South Australian Environment Protection Authority conducted a 10 week study in relation to noise complaints at the Waterloo wind farm in the Mid North of South Australia. The study involved the use of diaries by 28 households located around the wind farm and noise loggers at 6 dwellings. In analyzing the residents' diaries, the Environment Protection Authority focussed on "noise" events and disregarded comments or observations by the residents which related to other forms of disturbance such as sensation. In the light of the Schomer principle presented in the 2017 ASA meeting in Boston, the 2013 Waterloo study data has been revisited to include an examination of the Power Output and WAV files associated with periods where residents have reported high levels of disturbance.

4:45

4pNS12. Politics, regulatory, and acoustic failures. Natalie J. Squire (Acoustar, 18 Macs Rd., Buninyong, VIC 3357, Australia, wnsquire@bigpond.com.au)

There have been three senate inquiries in recent years, all of which have called for regulatory reform. Noise related nuisances and associated amenity detriments are very often attributed to the operation of nearby wind turbines. In many circumstances, disturbances are reported despite the reports of acoustic experts concluding that the wind farm is operating in compliance with the noise limits conditioned in relevant planning approvals. The current practise whereby wind farm operators directly engage acousticians requires review. This is particularly so in the state of Victoria where under-resourced municipal councils, usually without the technical capacity to understand complex acoustic issues, are now responsible for wind farm permit regulation and enforcement—and a state where the EPA plays no formal role in the assessment of noise or mitigation of complaints of noise pollution emitted from a wind farm. The societal benefits of independent acoustic reporting and the importance of peer review is discussed in relation to the Waubra and Cape Bridgewater wind farms.

5:00

4pNS13. Startle reflex and sensitisation—How are these biological phenomena relevant to wind turbine noise exposure? Sarah E. Laurie (Waubra Foundation, PO Box 7112, Banyule, Melbourne, VIC 3084, Australia, sarah@waubrafoundation.org.au), Steven E. Cooper (The Acoust. Group, Lilyfield, NSW, Australia), and Robert Thorne (Acoustar, Brisbane, QLD, Australia)

Recent laboratory and field research has identified strong amplitude modulation as a trigger for sleep disturbance via acute physiological stress events. Reported, observed, and objectively recorded sudden increases in heart rate as part of a "flight fight response" during both day and night time noise exposure suggests that direct stimulation of the sympathetic nervous system via the startle reflex response may be involved. Mammalian field research has demonstrated that repeated elicitation of the acoustic startle reflex leads to observed sensitisation. Sensitization is also observed in Individuals chronically exposed to amplitude modulated industrial noise from sources including wind turbines. Relevant existing scientific literature, and examples of these events will be discussed.

5:15

4pNS14. Noise prediction of axial fan duct using a lattice Boltzmann approach. Kentaro Hayashi (Res. & Innovation Ctr., Mitsubishi Heavy Industries,Ltd., 5-717-1, Fukahori-machi,Nagasaki, Nagasaki 850-0392, Japan, kentaro1_hayashi@mhi.co.jp)

Large axial fans in fire power plants are noise source of the plants, they are required to reduce the noise level. Fan noise propagates in a fan duct, and since it is emitted to the exterior, silencers are used as countermeasure in general. It is important for noise reduction to predict noise generation and noise propagation in fan ducts. The Lattice-Boltzmann Method (LBM) based approach was applied for the unsteady simulation of axial fan and its duct to predict fan noise generation and sound propagation simultaneously. The LBM has minute numerical viscosity, and is suitable for the analysis of the sound propagation and the flow field. In this research, aeroacoustic analysis by using applicable analysis model in the design phase has been conducted and validated compared to experimental results. The results shows good agreement with experiment within 3dB at overall value and predicted spectra are also compared with experimental results.

Session 4pPA

Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Acoustics of Detecting Gravitational Waves using LIGO

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Kenneth E. Gilbert, Cochair
Physics/NCPA, University of Mississippi, P.O. box 35, 1703 Hunter Road, Thaxton, MS 38871

Chair's Introduction—2:30

Invited Papers

2:35

4pPA1. Gravitational wave sources and data analysis. Benjamin Owen for the LIGO Scientific Collaboration and the Virgo Collaboration (Phys. and Astronomy, Texas Tech Univ., Box 41051, Lubbock, TX 79409-1051, benjamin.j.owen@ttu.edu)

What we look for affects how we look for it. I summarize current and predicted sources of gravitational waves, how the physics affects the signals, and how the signal morphologies affect analysis techniques. Expecting long and short signals, well modeled signals and unknown signals, means that gravitational waves use a variety of techniques familiar from acoustics, from matched filtering to wavelets and cross-correlation of data streams. These tell us today about black holes, and one day will tell us about neutron stars, cosmic explosions, and perhaps even stranger things.

3:05

4pPA2. A technical overview of the LIGO detectors. Adam Mullavey (LIGO Livingston Observatory, 19100 LIGO Ln., Livingston, LA 70754, amullavey@ligo-la.caltech.edu)

The LIGO detectors are a technological marvel, the culmination of ~50 years of research and development, that have opened up a new window on the universe by making the first detections of gravitational waves. The LIGO detectors can measure strain caused by gravitational wave perturbations down to 10^{-23} (at their most sensitive frequency band). In this talk, I will give an overview of the techniques that have allowed us to reach such incredible precision, and where possible relate said techniques to the field of acoustics.

3:35

4pPA3. Gravitational-wave detectors: Upgrades and new facilities. Katherine Dooley (Phys. and Astronomy, Univ. of Mississippi, PO Box 1848; 108 Lewis Hall, University, MS 38677, kldooley@olemiss.edu)

The LIGO observatories have detected gravitational waves, but even so there's much that can be done to further improve their sensitivity. I will describe plans to upgrade the existing Advanced LIGO detectors as well as design studies for new detectors in brand new facilities.

4:05–4:35 Panel Discussion

Session 4pSC

Speech Communication: Speech Perception and Word Recognition (Poster Session)

Mark VanDam, Chair
Speech & Hearing Sciences, Washington State University, PO BOX 1495, Spokane, WA 99202

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

4pSC1. Vowel intelligibility in clear speech produced by electrolaryngeal speakers. Steven R. Cox, Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., Hy Weinberg Ctr. 136, Garden City, NY 11530, scox@adelphi.edu), and Philip C. Doyle (Health and Rehabilitation Sci., Western Univ., London, ON, Canada)

This study assessed vowel intelligibility when electrolaryngeal (EL) speakers were instructed to use clear speech (CS). Eighteen consonant-vowel-consonant words containing /i/, /ɪ/, /ɛ/, and /æ/ were spoken by 10 laryngectomees in habitual speech (HS) and CS conditions. A total of 4,320 words across both speech conditions were recorded and then transcribed by 12, naïve listeners. Results indicate that vowel intelligibility was 85.4% (*range* = 77.8% to 90.6%) when EL speakers used HS compared to 82.7% (*range* = 75.0% to 92.8%) in CS. This finding suggests that CS does not facilitate improved vowel intelligibility for EL speakers. Future research will seek explanations for the lack of a CS benefit by examining the acoustic changes that occur when EL speakers use CS.

4pSC2. Phonological and auditory context effects in the perception of synthetic liquid-plus-stop clusters. Terrance M. Nearey and Benjamin V. Tucker (Linguist, Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 0A2, Canada, tnearey@ualberta.ca)

In experiments with English synthetic stops in the /VCCV/ disyllables /arga, alga, arda, alda/, listeners give more /-da/ responses after /ar-/ syllables [see, e.g., A. Lotto and K. Kluender, Percept. Psychophys. **60**, 602–619 (1998)]. The importance of general auditory contrast compared to more specific phonetic mechanisms is a key issue in the literature. In three experiments that used factorially crossed F3 continua for both the liquid /r-l/ and for the stop /g-d/, we recorded responses to all four disyllables (4 alternative forced choice). Experiments 1 and 2 both showed that categorizing the liquid as /r/ biases stop responses toward /d/. However, Experiment 1, which involved a 4-step extreme-/r/ to extreme-/l/ continuum, showed effects consistent with auditory contrast of the F3 values across the stop. Experiment 2 focused on the ambiguous region of the /l-r/ continuum [sampled in 7 steps]. While a clear effect of the [-rd-] phonological bias remained, there was little reliable evidence of contrast effects in F3 across the stop gap. Here, we report analyses of a new larger ($n > 60$) Experiment 3 with more stimuli (70 compared to less than 50 each) that subsumes the ranges of the previous two experiments. We evaluate the hypothesis (among others) that the auditory-contrast-like effect is confined largely to stimuli with relatively low F3 at the offset of the /VC-/ syllable.

4pSC3. Individual differences in cue weights are correlated across contrasts. Meghan Clayards (McGill Univ., 1085 Ave. Dr. Penfield, Montreal, QC H3A 1A7, Canada, meghan.clayards@mcgill.ca)

When listeners make judgments about phonological contrasts, they integrate different acoustic dimensions putting more weight on some than others. Individuals differ in how they weight different cues. Schultz, Francis & Llanos (2012) found that weights for VOT and f0 as cues to initial stop voicing in English are weakly positively correlated across individuals. We ask whether this is specific to VOT and f0 or a more general property of cue weighting across individuals. Secondly, across contrasts do the same listeners have stronger cue weights? 43 listeners performed a 2AFC task for four sets of minimal pairs that each varied orthogonally in two dimensions. All heard *bet-bat* and *Luce-lose* (vowel spectral quality vs. duration) and *bog-dog* (burst spectrum vs. formant transitions). 24 participants also heard *sock-shock* (sibilant spectrum vs. formant transitions) and the other 19 heard *dear-tear* (VOT vs. f0). Cue weights were fit with random slopes in a logistic regression for each minimal pair. Weights were positively correlated across individuals both within and across contrasts for *bet-bat, Luce-lose*, and *sock-shock*. *Bog-dog* and *dear-tear* had less consistent results. Overall this indicates that some individuals are better able to extract and use acoustic-phonetic information across different acoustic dimensions.

4pSC4. Intelligibility of sinewave consonants. James Hillenbrand and Michael J. Clark (Western Michigan Univ., 1903 W Michigan Ave., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

A good deal of experimental work has assessed the intelligibility of sinewave speech (SWS), synthesized by mixing sinusoids that follow the formants of natural utterances. While SWS is clearly intelligible at some level, most SWS work has been conducted using sentences, whose intelligibility is affected by many factors in addition to those related to recognition at the phonetic level. Earlier work [Hillenbrand *et al.*, *J. Acoust. Soc. Am*, **129**, 3991–4000] measuring the intelligibility of SWS vowels in isolated syllables reported an identification rate of 55%, far above chance but ~40 percentage points lower that of the original signals. The present work tested the intelligibility of SWS versions of 23 consonant types in CV and VCV syllables with three vowel types ([ɑ i u]) spoken by one man and one woman. SW signals were generated from unedited envelope peaks rather than formants. Intelligibility averaged across 59 listeners was 59%, with large variability across both listeners ($sd = 9.9$) and, especially, consonant type ($sd = 23.4$). Recognition at the feature level was 65.5% for place, 78.4% for

manner, 90.0% for voicing. Voicing results are especially striking given that SWS is fully aperiodic, and the fact that the representation of spectral shape in SWS is necessarily coarse.

4pSC5. Computational modeling of human isolated auditory word recognition using DIANA. Filip Nenadic (Linguist, Univ. of Alberta, Edmonton, AB, Canada), Louis ten Bosch (Radboud Univ., Nijmegen, Netherlands), and Benjamin V. Tucker (Linguist, Univ. of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

In recent years, computational modeling has proved to be an essential tool for investigating cognitive processes underlying speech perception (see, e.g., Scharenborg & Boves, 2010). Here we address the question of how an end-to-end computational model that uses the acoustic signal as input simulates behavioral responses of actual participants. We used the Massive Auditory Lexical Decision (MALD) database recordings comprising of 26,800 isolated words produced by a single male native speaker of English. MALD response data came from 232 native speakers of English, with each participant responding to a subset of recorded words in an auditory lexical decision experiment (Tucker *et al.*, submitted). We applied DIANA, a recently developed end-to-end computational model of word perception (Ten Bosch *et al.*, 2013; Ten Bosch *et al.*, 2015) to model the MALD response latency data. DIANA is a model that takes in the acoustic signal as input, activates internal word representations without assuming prelexical categorical decision, and outputs estimated response latencies and lexicality judgements. We report the results of the participant-to-model comparison, and discuss the simulated between-word competition as a function of time in the DIANA model.

4pSC6. Asymmetric discrimination of phonetically incongruent audiovisual vowels. Matthew Masapollo (Dept. Cognit., Linguistic & Psychol. Sci., Brown Univ., 190 Thayer St., Providence, RI 02912, matthew_masapollo@brown.edu), Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), James Morgan, Lauren Franklin (Dept. Cognit., Linguistic & Psychol. Sci., Brown Univ., Providence, RI), and Lucie Ménard (Dept. of Linguist, Univ. of Montreal at PQ, Montreal, QC, Canada)

Masapollo, Polka, and Ménard (2017) recently reported a robust directional asymmetry in visual vowel perception: perceivers discriminated a change from an English /u/ viseme to a French /u/ viseme significantly better than a change in the reverse direction. This asymmetry parallels a frequent pattern in auditory vowel perception that points to a universal bias favoring "focal" vowels, whose distal articulatory gestures result in the convergence of adjacent formant frequencies. In the present study, we investigated how the integration of acoustic and visual speech cues influences these directional effects in bimodal vowel perception. In AX discrimination tests in which acoustic and visual cues were phonetically-incongruent, subjects showed an asymmetry, comparable to that found with unimodal stimuli, when there was an acoustic change in focalization, with visual focalization held constant. These findings indicate that acoustic cues are capable of inducing an asymmetry even when there is no cross-modal correspondence between the acoustic and visual channels. We are currently examining whether a similar asymmetry also emerges when there is a visual change in focalization, with acoustic focalization held constant. By comparing different types of audio-visual incongruence, we will be able to examine whether one sensory channel has more perceptual potency than the other.

4pSC7. Individual differences in perceptual adaptation to phonetic categories: Categorization gradiency and cognitive abilities. Donghyun Kim, Meghan Clayards (Linguist, McGill Univ., 1085 Dr. Penfield, Rm. 111, Montreal, QC H3A 1A7, Canada, heydonghyun@gmail.com), and Eun Jong Kong (Dept. of English, Korea Aerosp. Univ., Goyang-si, Gyeonggi-do, South Korea)

We examine whether listeners flexibly adapt to unfamiliar speech patterns such as those encountered in foreign-accented English vowels. In these cases, the relative informativity of acoustic dimensions (spectral quality vs. duration) can be changed such that the most informative dimension (spectral quality) is no longer informative, but the role of the secondary cue (duration) is enhanced. We further test whether listeners' adaptive strategies are related to individual differences in utilizations of secondary cues (measured by categorization gradiency) and cognitive abilities. Native English listeners (N = 36) listened to continuum of vowels /ɛ/ and /æ/ (as in *head* and *had*) varying spectral and duration values to complete a perceptual adaptation task, a visual analogue scaling (VAS) task, and were given cognitive ability tasks examining executive function capacities. Results showed that listeners mostly used spectral quality to signal vowel category at baseline, but rapidly adapted by up-weighting reliance on duration when spectral quality was no longer informative. The VAS task showed substantial individual differences in categorization gradiency with more gradient listeners using a secondary cue more, but gradiency was not linked to degree of adaptation. Finally, results of cognitive ability tasks revealed that individual differences in inhibitory control, but not the other cognitive abilities, correlated with the amount of perceptual adaptation.

4pSC8. Working memory capacity and lexical knowledge in perceptual restoration of interrupted speech. Naveen K. Nagaraj (Audiol. and Speech Pathol., Univ. of Arkansas Medical Sci., 2801 S. University, Ste. 600, Little Rock, AR 72204, nknagaraj@uams.edu) and Beula Magimairaj (Commun. Sci. and Disord., Univ. of Central Arkansas, Conway, AR)

Role of working memory capacity (WMC) and lexical knowledge in perceptual restoration (PR) of missing speech was investigated using the interrupted speech perception (ISP) paradigm. 75 young normal hearing listeners' speech identification was measured using low-context sentences interrupted by silence and three noise at 1.5 Hz. Noise conditions created by manipulating the spectro-temporal content of filler noise were as follows: (1) low frequency (LF) speech shaped noise (SSN), (2) temporal fine structure filled (TFSf) noise consisting of LF TFS from the missing speech, and (3) temporal envelope filled (TEf) noise consisting of LF TE extracted from the missing speech. WMC was measured using verbal reading span and visuospatial symmetry span. Lexical knowledge was assessed using standard vocabulary and meaning from context tests. We hypothesized that during noise filled ISP conditions WM mechanism is crucial for retrieving and integrating relevant information from long-term memory. Results showed that ISP was better for SSNf than other conditions tested. Both lexical knowledge and verbal WMC explained unique variance in SSNf, but were unrelated to silent gated condition. It was only, lexical knowledge that uniquely predicted PR for TFSf and TEf conditions. Findings in general suggest that PR of filled interrupted speech depends crucially on individuals' lexical knowledge.

4pSC9. Delayed effects of speech and non-speech stimuli on sibilant categorization. Eleanor Chodroff (Linguist, Northwestern Univ., 2016 Sheridan Rd., Chicago, IL 60208, eleanor.chodroff@northwestern.edu) and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD)

Adaptation to the speech of a novel talker can involve at least two types of mechanism: phonetic learning (e.g., Samuel & Kraljic, 2009) and spectral contrast (e.g., Lotto & Kluender, 1998). While phonetic learning persists over long time periods, auditory effects such as spectral contrast have been demonstrated to occur for sounds that are temporally adjacent or separated by brief delays (e.g., 1.3s; Holt, 2005). The present study examined whether phonetic and auditory mechanisms can be distinguished by introducing a substantially longer delay between exposure and test. Five exposure conditions were examined across participants: exposure to syllables beginning with a relatively high or low COG [z], exposure to white noise matched in LTAS to the high or low COG [z]-initial syllables, and a baseline (no exposure) condition. Following exposure, participants performed a one-back image repetition detection task for 15 minutes, and then categorized members of the same 10-point [ʃ]-[s] continuum over 6 blocks. In comparison to baseline, exposure to speech had a significant and expected influence on sibilant categorization (e.g., the [ʃ]-[s] boundary shifted toward [s] following high COG [z] exposure). Speech and non-speech exposures were compared to determine whether spectral contrast is a viable explanation for long-lasting sibilant adaptation.

4pSC10. Effect of musical experience on learning to understand vocoded speech. Kieran E. Laursen (Dept. of Psychol., Lawrence Univ., 711 E. Boldt Way, Appleton, WI 54911, kieran.e.laursen@lawrence.edu), Iain C. Williams (Psychology, Univ. of North Carolina, Wilmington, NC), Tahnee Marquardt (Oxford Mindfulness Ctr., Univ. of Oxford, Oxford, United Kingdom), Sara L. Prostko, and Terry L. Gottfried (Psychology, Lawrence Univ., Appleton, WI)

This study explores whether musicians have an advantage over non-musicians in processing and comprehending 8-channel vocoded speech, spectrally degraded speech that imitates cochlear implant output (see Loebach, Bent, & Pisoni, 2008). Musicians and non-musicians completed a pre-test in which they were asked to transcribe a number of vocoded sentences and words. Afterwards, they completed training on either vocoded or natural stimuli. After training, participants completed post-tests during which they transcribed vocoded speech, including items from the original test and from new speakers and 4-channel vocoded stimuli. In preliminary studies, musicians showed no advantage over non-musicians, and contrary to expectations, participants in natural-speech training condition improved more from pre- to post-test vocoded word recognition than participants in the vocoded training condition. Current studies are examining the relation of rhythmic and melodic perception (Musical Ear Test, Wallentin *et al*., 2010) to performance on the vocoded speech learning; the vocoded and natural speech training tasks are also compared to a control training task (testing sine-wave tone detection). Results of these tests will be used to evaluate the extent to which musical ability and training may be related to perception of degraded speech.

4pSC11. The role of familiarity in audiovisual speech perception. Chao-Yang Lee, Margaret Harrison (Commun. Sci. and Disord., Ohio Univ., Grover W225, Athens, OH 45701, leec1@ohio.edu), and Seth Wiener (Modern Lang., Carnegie Mellon Univ., Pittsburgh, PA)

Speech perception involves processing a spoken message's linguistic content and information about the talker's voice carrying the message. Speech perception is also affected by visual information, e.g., the McGurk effect. Familiarity with a talker's voice facilitates auditory speech perception (Nygaard & Pisoni, 1998), but it is not clear whether visual familiarity with a talker's face affects audiovisual speech perception (Walker, Bruce, & O'Malley, 1995). In this study we investigated how visual familiarity with a talker affects the perception of the English sibilants /s/ and /ʃ/, which involve visible lip-rounding contrasts. Participants identified syllable-initial sibilants from stimuli that were audio-only, visual-only, audiovisual-congruent, or audiovisual-incongruent (e.g., audio "save" paired with visual "shave"). We also examined whether visual familiarity affects the occurrence of the McGurk effect. Participants identified syllable-initial stops from syllables that were audiovisual-congruent or incongruent (e.g., audio /bɑ/ paired with visual /gɑ/). The results indicated that participants familiar with the talker identified sibilants faster in all conditions and more accurately in the visual-only condition; no accuracy difference was found between the two groups in the other conditions or in the number of McGurk responses. These results are discussed in the context of processing intersensory information.

4pSC12. Examining the effect of first formant transition on perception of dynamic spectral change in the second formant. Amie Roten and Michelle R. Molis (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland HCS, 3710 SW US Veterans Hospital Rd., P5-NCRAR, Portland, OR 97239, amie.roten@va.gov)

Speech perception studies using methods such as formant-flattening and the silent-center paradigm have demonstrated that vowel-inherent spectral change (VISC) in speech-like stimuli adds significant information aiding in accurate phoneme identification. For listeners to reliably utilize this cue, formant extent thresholds must be smaller than transitions occurring in natural syllables. Although natural speech always involves movement of multiple formants over time, limited studies have determined formant transition thresholds in the context of competing formant movement. The purpose of this study was to begin to examine these interactions by determining the effect of F1-region transitions on perception of change in F2 (ΔF2). Listeners were presented 120 ms, two-formant stimuli approximating a vocoid F1/F2 configuration using a 4-interval, 2AFC, two-down/one-up adaptive paradigm. The first formant pivoted around a 500 Hz center frequency, and the second around 1500Hz. Thresholds for detection of both upward and downward ΔF2 were determined in the context of a flat F1, and an F1 transitioning linearly up or down by 1 ERB. Differences in performance were observed depending on F1 context, as well as direction of ΔF2, which may have bearing on speech perception ability. [Work supported by NIH/NIDCD.]

4pSC13. Study on interactions between voicing production and perception using auditory feedback paradigm. Shunsuke Tamura, Miduki Mori, Kazuhito Ito, Nobuyuki Hirose, and Shuji Mori (Kyushu Univ., 744 Motooka, Nishi-ku, Fukuoka, Japan, Rm. 827, 8th Fl., West Zone II Bldg., Kyushu University Ito Campus, Fukuoka 819-0395, Japan, tamuras@cog.inf.kyushu-u.ac.jp)

A previous study reported that perturbed auditory feedback affected voicing production [Mitsuya, MacDonald, and Munhall (2014). J. Acoust. Soc. Am., **135**, 2986–2994]. In this study, we investigated whether perturbed auditory feedback would also affect voicing perception. Eighteen native Japanese speakers participated in the experiment. Half of the participants performed an auditory feedback task in which a syllable sound /da/ was presented simultaneously with the participant's utterance of /ta/. The other participants did a passive listening task in which participants heard a syllable sound /da/ without the utterance. Before and after each task, participants performed a speech production task of /da/-/ta/ and a speech identification task of /da/-/ta/ continuum stimuli varying in voice-onset time (VOT). Results showed that perturbed auditory feedback lengthened the VOT of /ta/ production, whereas passive listening did not affect voicing production. Regarding voicing perception, passive listening shortened the VOT boundary of /da/-/ta/, which may be due to selective adaptation. On the other hand, perturbed auditory feedback did not vary the boundary. One interpretation of these results is that the effects of voicing production modulation on voicing perception can be cancelled out by selective adaptation, which may have occurred by listening to a syllable sound /da/ during auditory feedback task.

4pSC14. Segmenting words from bilingual speech: Evidence from 8- and 10-month-olds. Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, QC H3A 1G1, Canada, adriel.orena@mail.mcgill.ca) and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Previous studies show that young monolingual infants use language-specific cues to segment words from multiword utterances in their native language. However, little is known about how infants deal with the segmentation challenge in bilingual environments. Here, we examined the word segmentation abilities of young infants in a mixed dual-language task. Infants were familiarized with an English-French passage containing a target word in each language, and were then tested on their recognition of those target words. Results confirm that 8-month-old monolingual infants show language-specific patterns in word segmentation: English- and French-monolingual infants segmented in their native language, but not in the other unfamiliar language. As a group, 8- and 10-month-old bilingual infants showed positive evidence of segmentation in both of their native languages. However, closer inspection of the data suggests that bilingual infants are only able to segment in the language in which they receive more exposure. Taken together, these results suggest a dose-response relationship between speech input and word segmentation: that is, more input in a language gives infants more opportunities to learn about how word boundaries are denoted in that language. This study also addresses the possible roles of attention and code-mixing on the development of word segmentation abilities.

4pSC15. Effect of speaker's age on perceivers' ability to predict rounding from AV speech. Melissa Redford (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu) and Jeffrey Kallay (Univ. of Oregon, Eugene, Ohio)

We have developed a noninvasive method that synthesizes the complex activity of speaking into a single measure to investigate the scope of anticipatory behavior as this relates to units of execution (Redford, Bogdanov, Vatikiotis-Bateson, 2016). The method leverages work in audio-visual speech perception to identify degree of anticipatory coarticulation at specific temporal locations, manipulated using a gating paradigm. The aim here was to test whether the method was robust to group differences in speakers' age. Twenty-four college-aged adults judged the presence or absence of rounding in minimal pair sentences produced by 3 adults and 3 five-year-old children based on their AV-recorded speech, gated at different distances from the target rounded/unrounded vowel. Perceivers' ability to correctly detect an upcoming rounded vowel varied as a function of the prevocalic consonant's coarticulatory resistance and as a function of distance from target. There was no main effect of age group on perceivers' accuracy. Age group did however interact with the other factors in that rounding was detected earlier in children's speech compared to adults' speech for one of three prevocalic consonants. The results confirm the promise of our method for studying developmental changes in linguistic chunks planned for execution. [Work supported by NICHD # R01HD087452.]

4pSC16. Do older and younger adult listeners show a "clear speech benefit" for speech produced by older adult talkers? Valerie Hazan and Outi Tuomainen (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

This study investigates whether a "clear speech benefit" is obtained for speech produced by older (OA) talkers and younger adult (YA) controls in a clear speaking style when heard in babble noise. The speech materials were recorded while OA and YA talkers read BKB sentences to a YA partner who repeated the sentence while hearing normally (NORM) or with a simulated hearing loss (HLS). The HLS condition naturally induced clear speech adaptations. 128 BKB sentences from 4 YA and 4 OA talkers (NORM, HLS) matched on a range of metrics were used in an adaptive listening test tracking the signal-to-noise ratio corresponding to 67% intelligibility. Listeners were 71 native British English listeners: 24 YA (M = 25.2 yrs), 27 OA-NH with normal hearing (M= 71.8), 20 OA-HL with presbyacusis (M= 73.7). Speech perception in noise was hardest for OA listeners, especially from OA-HL. SNR thresholds were significantly lower for YA than for OA voices. The clear speech benefit for HLS speech was only significant for YA voices for all listener groups. In summary, OA talkers were less intelligible than YA talkers regardless of listener age; clear speech adaptations by OA talkers did not result in enhanced intelligibility for OA or YA listeners.

4pSC17. Inhibitory and lexical frequency effects in younger and older adults' spoken word recognition. Sarah Colby (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, QC H3A 1G1, Canada, sarah.colby@mail.mcgill.ca), Victoria Poulton, and Meghan Clayards (Linguist, McGill Univ., Montreal, QC, Canada)

Older adults are known to have more difficulty recognizing words with dense phonological neighbourhoods (Sommers & Danielson, 1999), suggesting an increased role of inhibition in older adults' spoken word recognition. Revill and Spieler (2012) found that older adults are particularly susceptible to frequency effects, and will look more to high frequency items compared to younger adults. We aim to replicate and extend the findings of Revill and Spieler (2012) by investigating the role of inhibition along with frequency for resolving lexical competition in both older and younger adults. Older (n = 16) and younger (n = 18) adults completed a visual word paradigm eyetracking task that used high and low frequency targets paired with competitors of opposing frequency, and a Simon task as a measure of inhibition. We find that older adults with poorer inhibition are more distracted by competitors than those with better inhibition and younger adults. This effect is larger for high frequency competitors compared to low. These results have implications for the changing role of inhibition in resolving lexical competition across the adult lifespan and support the idea that decreased inhibition in older adults contributes to increased lexical competition and stronger frequency effects in word recognition.

4pSC18. Speech recognition and word learning in 24-month-olds: The roles of non-native speech and familiar words. Cynthia P. Blanco and Sandra R. Waxman (Psychology, Northwestern Univ., 2029 Sheridan Rd., Evanston, IL 60208, cynthiapblanco@gmail.com)

After 18 months of age, infants' lexical representations are sufficiently flexible to recognize acoustically unfamiliar productions as variants of familiar words (Best *et al*., 2009; Mulak *et al*., 2013). For novel words, 24-month-olds still have trouble generalizing from native to non-native pronunciations, although exposure to the accent improves recognition (Schmale *et al*., 2011, 2012; White & Aslin, 2011). In the present study, we tested how quickly 24-month-olds could use exposure and familiar words to learn the meanings of novel words, on-line, when listening to Spanish-accented speech. In the exposure phase, twenty-four-month-olds heard a novel word embedded in a dialogue that either contained linguistic cues to the referent's animacy (*The vep is eating*) or was uninformative (*The vep is right here*). Infants then saw two pictures at test, one animate and one inanimate, and were asked to find the *vep*. Looking time to the two potential referents here was compared to performance with native-accented speech (Ferguson *et al*., under review). Infants hearing Spanish-accented speech struggled to learn the referents of novel words, but differences also emerged between the accent conditions for familiar target words. Infants are slower to access the meanings of familiar words when hearing non-native speech.

4pSC19. Does the developing lexicon constrain infants' discrimination of English vowels? Megha Sundara (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, megha.sundara@humnet.ucla.edu)

Infants initially discriminate native and non-native contrasts. With perceptual reorganization within the first year of life, discrimination of non-native contrasts reduces while the discrimination of native contrasts improves. Several researchers have proposed distributional learning as a domain-general mechanism by which infants' acquire phonetic categories (e.g., Saffran *et al*., 1999; Maye *et al*., 2002). Recent proposals, however, argue for an interactive mechanism where learning words concurrently, supplements distributional learning of phonetic categories (e.g., Swingley, 2009; Feldman *et al*., 2013). Not only is this interactive mechanism available during the first year of life, its computational implementations outperform ones based on distributional learning alone (Feldman *et al*., 2013a; Feldman *et al*., 2013b). We evaluated predictions of the interactive and distributional learning models against infant discrimination data. Data from 4-, and 8-month-olds, learning (a) only English (b) only Spanish or (c) Spanish and English, tested on English (i) /e- ɛ/ (previously published in Sundara & Scutellaro, 2011), (ii) /i- ɪ/ and (iii) /e- ɪ/, were compared. All stimuli were produced by multiple female talkers from the Hillenbrand corpus; infants were tested using a visual fixation procedure with a habituation criterion of 50%. Our results are consistent with infants using a distributional not interactive learning mechanism.

4pSC20. Representations of speech signals recorded through a dynamic periphery inspired by horseshoe bat biosonar. Alexander Hsu (T. J. Watson Res. Ctr., IBM, Yorktown Heights, NY), Anupam Kumar Gupta, Rolf Müller (Virginia Tech, Blacksburg, VA), Xiaodong Cui, Kartik Audhkhasi, and Jin-Ping Han (T. J. Watson Res. Ctr., IBM, 1101 Kitchawan Rd., Yorktown Heights, NY 10598, hanjp@us.ibm.com)

Horseshoe bats have to navigate through complex environments such as dense forests and structure-rich vegetation relying on input from their highly sophisticated biosonar systems. One of the key components of these bats' ability to obtain high-quality acoustic information is to alter the shape of their outer ears rapidly. In prior work, the authors have shown that by mimicking the horseshoe bat rapid ear movements, a bat-inspired robotic dynamic periphery for recording speech signals could enhance speech recognition for limited dataset and also provide estimates for the speaker's direction along with speech recognition. In our current study, we continued to investigate how speech datasets processed by the dynamic periphery may be enhanced compared to a reference by extracting acoustical features through Mel frequency cepstral coefficient (MFCC) transform, Lyon's cochlear bandpass filters, and a neural spike representation, respectively. This study aims to characterize the detailed acoustical differences and quantify the improved speaker intelligence with noise robustness through the dynamic periphery. The ultimate goal of this research is to identify a signal representation that is well suited to capitalize on the time-variant properties of the biomimetic recording periphery and make the dynamic information-bearing features accessible for the classification stages.

4pSC21. Filtered and unfiltered sentences produce different spectral context effects in vowel categorization. Christian Stilp and Ashley Assgari (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech perception is heavily influenced by surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, this can produce *spectral contrast effects (SCEs)* that bias categorization of later sounds. For example, when context sounds have a low F1 bias, listeners report more high F1 responses to the target vowel, and vice versa. SCEs have been demonstrated using a variety of approaches, but most often the context was a single sentence filtered two ways (e.g., low F1 bias, high F1 bias) to introduce spectral properties that biased speech categorization. This maximizes acoustic control over stimulus materials, but vastly understates the acoustic variability of speech. Here, vowel categorization was examined following context sentences that naturally possessed desired spectral properties without any filtering. Sentences with inherent low-F1 or high-F1 peaks in their long-term spectra were presented before a target vowel (/ɪ/-/ɛ/). Filtered sentences with equivalent spectral peaks were included as controls. Across several experiments, as spectral peak magnitudes in filtered and unfiltered contexts increased, SCE magnitudes increased linearly. However, unfiltered sentences produced smaller and more variable SCEs than filtered sentences. Results raise important questions about how closely laboratory studies of speech perception model processes in everyday listening.

4pSC22. Discrete ratings and dimensional judgments of emotional speech: A preliminary look at gender differences. Shae D. Morgan and Rebecca Labowe (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Two models of emotion identification and discrimination are not often compared and may show differing trends in how listeners rate emotional stimuli. The discrete or "basic" emotion model posits that emotions are categorical, and that listeners develop auditory prototypes of "basic" emotions (e.g., anger) based on their acoustic profile and past experiences. More complex emotions (e.g., frustration) fall under a "basic" category. The dimensional model examines emotions along continua of different emotional dimensions, such as activation/arousal and pleasantness/valence. The present study introduces the Morgan Emotional Speech Set and examines listener judgments of the stimuli in the corpus. The database consists of 2160 emotional speech sentences (90 sentences x 4 emotions x 6 talkers) produced by three male and three female actors for use in future studies. Each sentence was rated by 10 listeners (5 male, 5 female), who assigned an emotion category to each sentence and also rated each sentence by its activation (low to high) and pleasantness (very unpleasant to very pleasant). Discrete ratings will be compared with dimensional judgments made by listeners to examine intended and perceived emotional content in the database. A preliminary look at gender differences in the ratings and judgments will also be discussed, as male and female listeners may rate emotions differently for same- or different-sex talkers.

4pSC23. F0 as a cue to prosodic structure in voice onset time categorization. Jeremy Steffman (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jsteffman@g.ucla.edu)

This study investigates the effect of F0-based cues to English prosodic structure on the categorization of voice onset time (VOT) continua. Human perceptual systems are capable of extracting relevant phonetic and phonological information in spite of pervasive variation in the speech signal, with prosodic structure being one source of variation. For example, in English VOT in voiceless stops is robustly longer when intonational phrase (IP) initial, versus IP-medial. Kim & Cho (2013) found listeners compensate for perceived IP boundaries, requiring longer VOT to categorize sounds as voiceless when IP-initial. However, because the target sound was in an utterance-medial IP, compensation could have arisen due to the lengthened duration of the preceding IP-final syllable, as in speech rate normalization, instead of due to prosodic structure itself. Mitterer *et al*. (2016) found categorization shifts solely on the basis of duration, by rendering F0 contours prosodically ambiguous, yet it remains unknown what effect F0 has on categorization when it unambiguously cues prosodic structure. This study tests the hypothesis that compensation is triggered by more than duration normalization, by varying F0 cues independently from duration. Results will be discussed in terms of language-general auditory processing and top-down language-specific influences on segment categorization.

4pSC24. Perceived anger in clear and conversational speech: Effect of aging and hearing loss. Shae D. Morgan and Ashton Crain (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

Previous research suggests that young normal-hearing and older hearing-impaired adult listeners judge clear speech as sounding angry more often than conversational speech. Interestingly, older hearing-impaired listeners were less likely than young normal-hearing listeners to judge sentences as angry in both speaking styles. It was unknown, however, whether this difference in ratings of emotion were driven by the age or hearing status differences between the two groups. A secondary investigation showed that young adult listeners with a simulated hearing loss that matched the older hearing-impaired group rated emotions nearly identically to the young normal-hearing group, suggesting no effect of hearing loss on ratings of emotion. The simulated hearing loss failed to account for other auditory factors or psychological processes associated with aging that may have account for the group differences. The present study carried out the same emotional rating task using clear and conversational speech sentences (as used in the previous studies) on a group of older adults with clinically normal hearing to determine whether differences in anger (and other emotion) perception are driven by age differences, hearing loss, or other factors.

Session 4pSPa

Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity I

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Chair's Introduction—1:00

Invited Papers

1:05

4pSPa1. Exploiting natural surface noise and ships as sources of opportunity for passive bottom reflection-loss estimation. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

The marine ambient-noise field, including natural and anthropogenic sources, can be exploited to estimate the bottom reflection coefficient and the associated loss, which is an important contributor to the total transmission loss in shallow water scenarios. The physical nature of the natural surface-noise field is such that the bottom reflection coefficient can be estimated passively by beamforming on a vertical line array, if there are no localized sources near the array and the noise level generated by waves and wind is adequate. Recently, an algorithm for processing the noise generated by a single ship moving over a wide range of steering angles has been shown to extend the technique to this source of opportunity, even when the natural noise level is too low to produce any measurable loss. In this study, simulation and data from recent experimental campaigns (from arrays moored to the bottom or mounted on an autonomous underwater vehicle) are used to illustrate the technique, and investigate in particular both the effect on the performance of low natural noise levels, and the possibility of using more than one ship as source of opportunity.

1:25

4pSPa2. Headwaves in ocean acoustic interferometry. Martin Siderius (ECE Dept., Portland State Univ., P.O. Box 751, Portland, OR 97207, siderius@pdx.edu), Jie Li (Marine Physical Lab., Univ. of California, Scripps Inst. of Oceanogr., San Diego, CA), and Peter Gerstoft (Marine Physical Lab., Univ. of California, Scripps Inst. of Oceanogr., La Jolla, CA)

In ocean acoustic interferometry, signals measured on two or more receivers are cross-correlated to produces an estimate of the Green's function between these receivers. A "virtual refracted" wave is an early arrival in the time-domain Green's function estimate. This virtual refracted wave is a phenomenon that has been widely reported on in the seismic interferometry literature. This early arrival of energy is also referred to as spurious energy or non-physical arrivals. Although this can be interpreted as a virtual wave it can be a result of the physically propagating headwave. The headwave travels in the seabed and re-radiates into the water column and therefore has important information content such as the seabed sound speed. In seismic interferometry, active sources and horizontal arrays are used but the virtual refracted headwave phenomenon is also observable on vertical arrays and with passive measurements of ocean noise. The signal processing used is a generalization of passive fathometer processing which applies beamforming (including adaptive methods) to the array data. Modeling and experimental data will be presented to show the headwave can be observed and used to estimate the seabed sound speed. The generalized passive fathometer signal processing is compared to the seismic interferometry processing.

1:45

4pSPa3. Extracting tomographic arrival time information from ships of opportunity in the Santa Barbara channel using blind deconvolution. Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, J. Erskine Love Bldg., Rm. 131, Atlanta, GA 30332-0405, ncd001@lvc.edu), Kay L. Gemba, Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

This paper summarizes the ongoing investigations surrounding the use of a ray-based blind deconvolution algorithm to recover arrival time information from sources of opportunity, such as shipping vessels, recorded on vertical line arrays (VLAs) in ocean waveguides. The deconvolution is primarily performed by using an estimate of the unknown source phase, obtained through wideband beamforming, to essentially match filter the VLA recordings and recover the channel impulse response (CIR). This paper will focus on results from an experiment performed in 2016 in the Santa Barbara shipping channel (water depth ~550 m). Four VLAs, with both short (~15 m) and long (~56 m) apertures, were deployed between the north and south bound shipping lanes and continuously collected acoustic data during one week. With the ultimate goal of passive acoustic tomography in mind, this paper aims to discuss (1) the robustness of the algorithm to extract differential arrival times along VLA elements using ships as sources of opportunity, (2) the achievable accuracy of blind arrival time measurements in comparison to the time-of-flight precision required for tomographic inversions, and (3) the ideal parameters (e.g., frequency bandwidth, snapshot duration, beamforming methodology...) for which to perform this ray-based blind deconvolution method in SBC-like ocean environments.

2:05

4pSPa4. Differential arrival time accuracy using ships of opportunity. Kay L. Gemba, Jit Sarkar, Bruce Cornuelle (MPL/SIO, UCSD, Univ. of California, San Diego, La Jolla, CA 92093-0238, gemba@ucsd.edu), Nicholas C. Durofchalk, Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), William S. Hodgkiss, and William A. Kuperman (MPL/SIO, UCSD, La Jolla, CA)

Previously, we demonstrated using the Noise-09 data set that it is possible to estimate the relative channel impulse responses (CIR) over a 1.5 km transect independently at 3 vertical line arrays (VLAs). This vessel provided sufficient SNR and bandwidth for the uncertainty of the direct arrival time to reach 5 microseconds. The time evolution of the CIRs as the ship passed by was subsequently used to determine VLA hydrophone positions relative to a reference hydrophone with accuracy on the order of centimeters. An experiment was performed in the Santa Barbara Channel using four VLAs placed between the sea lanes of in- and outgoing shipping traffic. Ship tracks were obtained from the Automatic Identification System (AIS). We extend the previous processing to this data set and discuss the uncertainties of relative multi-path arrival-times and differential travel times between VLAs.

2:25–2:40 Break

2:40

4pSPa5. Active vs passive moving source tomography: Comparing results from the Santa Barbara Channel Experiment (SBCEx16) on sources of opportunity. Jit Sarkar, Bruce Cornuelle, Kay L. Gemba, William A. Kuperman, William S. Hodgkiss (Scripps Inst. of Oceanogr., UC, San Diego, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92093-0238, bsarkar@ucsd.edu), Karim G. Sabra (College of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Jeffery D. Tippmann, and Christopher M. Verlinden (Scripps Inst. of Oceanogr., UC, San Diego, La Jolla, CA)

An experiment was performed in the Santa Barbara Channel (SBCEx16) using four vertical line arrays (VLAs) of hydrophones placed midway between the in- and out-going shipping lanes of a maritime highway. The goal of this experiment was to study whether these sources of opportunity can be used for passive tomographic purposes. In addition to the continuous passive observations, active acoustic source-tows were conducted. The environment was monitored continuously throughout the course of the experiment using thermistor strings, and ship tracks were obtained from the Automatic Identification System (AIS). The results from both types of tomography are presented and compared.

3:00

4pSPa6. Using observations of commercial shipping from horizontal arrays deployed close to a shipping lane to map seabed properties. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

Several horizontal arrays were deployed on the seabed off the coast of Florida, outside Port Everglades. This is a busy site for cruise and container ships, commercial shipping obligated to transmit Automatic Identification System (AIS) information, indicating own lat-longs, as well as other parameters of their identity and motion. Acoustic data from these arrays and AIS transmissions were recorded for several days, capturing many passing ships that could then be used as sources at known locations for inversion of seabed properties. One of the arrays had geophones sensing three orthogonal vector components of particle velocity, as well as omni-directional pressure. This area is known to have a distressingly range-dependent seabed, making it difficult to successfully perform matched field processing, for example. We will present our processing of this data, including adaptive beamforming of the geophone data and methods that exploit beam migration patterns (in angle and time difference of arrival) to help isolate multipath arrivals, which are mapped via back-propagating rays to specific grazing angles and locations on the sea floor to provide measurements of bottom loss. The complete set of such observations from ship tracks at multiple frequencies, are incorporated into an inversion for seabed properties.

3:20

4pSPa7. Using vessel noise from a single hydrophone to estimate environmental properties. Graham A. Warner (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, graham.warner@jasco.com), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

This paper estimates seabed and water-column properties of a shallow-water site in the Chukchi Sea using vessel noise recorded on a single ocean-bottom hydrophone. A shallow-hazards seismic survey vessel transited with a fixed heading, passing within 200 m of the hydrophone. Sound pressure levels as a function of frequency and range are inverted using a trans-dimensional (trans-D) Bayesian approach to estimate range-independent environmental properties (sound-speed profile, water depth, and seabed geoacoustic properties). The trans-D inversion allows the data to determine the most appropriate environmental model parameterization in terms of the number of sound-speed profile nodes and subbottom layers. The inversion also estimates the vessel source levels, source depth, hydrophone height above the seabed, a range-correction factor, and error statistics, and provides uncertainty estimates for all model parameters and parameterizations. The sound-speed profile is found to be in good agreement with a measured profile, and the upper sediment-layer sound speed agrees with estimates from separate inversions of airgun and bowhead whale modal dispersion data. This method could be applied to monitor sound-speed profile changes over long time scales, e.g., using Automatic Identification System position data for vessels in a shipping lane.

3:40

4pSPa8. Echolocating dolphins as opportunistic signal sources for observing fine structure in the impulse responses of acoustic scatterers. Eric L. Ferguson, Stefan B. Williams (Australian Ctr. for Field Robotics, School of Aerosp. Mech. and Mechatronic Eng., University of Sydney, NSW, Australia), Craig T. Jin (Computing and Audio Res. Lab., School of Elec. and Information Eng., Sydney, NSW, Australia), and Brian G. Ferguson (Dept. of Defence, Defence Sci. and Technol., PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

Echolocating dolphins emit wideband acoustic signals as sequences of short duration pulses or *clicks*. Often, the waveforms of the clicks (especially during a buzz phase) have simple shapes that resemble a Ricker wavelet or a Gaussian modulated sinusoidal pulse. The main energy lobe of the pulse is of short duration (about 20 microseconds) with an inverted monopulse shape indicating a rarefaction (negative pressure pulse). An experimental bistatic active sonar system is configured in which a dolphin's biological sonar transmissions are coupled with a wide aperture receiving array. The acoustic transmissions of free-ranging Indo-Pacific bottlenose (*Tursiops aduncus*) dolphins are sampled every 4 microseconds at three collinear hydrophones having an interelement spacing of 14 m and located 1 m above the sea floor in shallow water of depth 20 m. Insonification of the seafloor and sea surface boundaries by the short-duration pulses reveals fine structure in the observed forward scattering impulse responses of the boundaries. Features (or highlights) in the forward scattering impulse responses are measured to within a fraction of a microsecond leading to precise localization of the scatterer using the modified wavefront curvature passive ranging method. Also, it will be shown that a similar result is observed for a scatterer in the water column that was moving rapidly with respect to the echolocating dolphin.

Session 4pSPb

Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Signal Processing Methods Exploiting the Information Content Provided by Sources of Opportunity II

Kay L. Gemba, Cochair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, CA 92037

Jit Sarkar, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Drive, Mail Code 0238, La Jolla, CA 92093-0238

Invited Papers

4:05

4pSPb1. Cascade of blind deconvolution and array invariant for robust source-range estimation. Hee-Chun Song, Chomgun Cho (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), gihoon byun, and Jea Soo kim (Korea Maritime and Ocean Univ., Busan, N/A, South Korea)

The array invariant proposed for robust source localization in shallow water is based on the dispersion characteristics in ideal waveguides. It involves conventional plane-wave beamforming using a vertical array, exploiting multiple arrivals separated in beam angle and travel time, i.e., beam-time migration. The approach typically requires either a short pulse emitted by a source or the Green's function that can be estimated from a probe signal to resolve distinct multipath arrivals. In this letter, the array invariant method is extended to unknown source waveforms by extracting the Green's function via blind deconvolution. The cascade of blind deconvolution and array invariant for robust source-range estimation is demonstrated using a 16-element, 56-m long vertical array at various ranges (1.5–3.5 km) from a towed source broadcasting broadband communication waveforms (0.5–2 kHz) in approximately 100-m deep shallow water.

4:25

4pSPb2. Acoustic interrogation of objects using signals from rotorcrafts. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

Rotorcraft generate acoustic waveforms that are repeatable, loud, and contains a large number of harmonics. They are a good source for generating step frequency-like waveforms that propagate over long distances and can easily be moved to interrogate new locations. One potential application of this source is a monostatic synthetic aperture sodar (SAS). A simple simulation was performed using back projection to generate SAS imagery. The movement of the helicopter generates the synthetic aperture, the signals generated by the main and tail rotors produce the waveform, and microphones mounted on the helicopter receive the reflected sound waves. The measured acoustic signals needs to be motion-compensated and focused, similar to the processing in synthetic aperture radar. A signal bandwidth of 170 Hz will generate an acoustic image with a downrange resolution (c/2B) of approximately 1 meter, and a 100 meter aperture will generate an acoustic image with a crossrange resolution (λ/d) of approximately 2 degrees for a center frequency of 100 Hz. Issues such as turbulence, saturation from self-noise, waveform estimation, acoustic reflectance of targets, range ambiguity, and wind need to be included in future simulations to obtain more realistic results.

4:45

4pSPb3. Using nonlinear time warping to estimate North Pacific right whale calling depths and propagation environment in the Bering Sea. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Julien Bonnel, Margaux Thieury (ENSTA Bretagne, Brest cedex 9, France), Aileen Fagan (U.S. Coast Guard Acad., New London, CT), Christopher M. Verlinden (SIO, UCSD, La Jolla, CA), Dana Wright, Jessica Crance, and Catherine L. Berchok (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., Seattle, WA)

Calling depth distributions and ranges are estimated for two types of calls produced by critically endangered eastern North Pacific right whales (NPRW) in the Bering Sea, using passive acoustic data collected with bottom-mounted single-hydrophone recorders in 50–90 m water depths. Nonlinear time resampling of 12 NPRW "upcalls" and 20 broadband "gunshots" typically isolated 3 to 4 individual mode arrivals below 200 Hz. Matched-mode processing (MMP) methods were used to remove the unknown source phase and amplitude structure, but incoherently averaging MMP ambiguity surfaces across frequency yielded many local minima when inverting for sediment properties. Instead, the ambiguity surfaces were plotted as a function of range and frequency, which revealed the type and degree of environmental mismatch present in initial waveguide models. This qualitative approach revealed the existence of large sound speed

gradients in the sediment, along with downward-refracting sound speed profiles during the summer months. Gunshot sounds were generally produced at a few meters depth, while upcall depths clustered between 10 and 25 m, consistent with previously published bioacoustic tagging results from North Atlantic right whales.

Contributed Papers

5:05

4pSPb4. Array invariant-based localization using ships of opportunity. Gihoon Byun, Jea Soo kim (Ocean Sci. and Technology-Korea Maritime and Ocean Univ., Korea Maritime Univ., Dongsam 2-dong, Yeongdo-gu, Busan 606-791, South Korea, gihoonbyun77@gmail.com), Chomgun Cho, Hee-Chun Song (Scripps Inst. of Oceanogr., Univ. of California, La Jolla, CA), and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

The array invariant (AI) proposed for robust source-range estimation with minimal knowledge of the environment in shallow water is based on the dispersion characteristics in ideal waveguides. This approach involves plane wave beamforming, utilizing coherent multiple arrivals separated in beam angle and travel time, referred to as “beam-time migration.” To resolve multipath arrivals in beam-time domain, AI requires either an impulsive source or Green’s function typically estimated from a known probe signal. For unknown source waveforms, it is possible to estimate the Green’s function using a ray-based blind deconvolution (RBD) which also utilizes simple conventional beamforming. Recently, the cascade of RBD and AI has been demonstrated for a towed source at 50-m depth broadcasting communication waveforms [*J. Acoust. Soc. Am.* **141**, 3270–3273 (2017)]. Rather than the towed source, this study focuses on the feasibility of tracking a ship radiating random and anisotropic noise. The combination of RBD and AI is demonstrated to localize a ship of opportunity (200–900 Hz) along a track at ranges of 1.8–3.4 km and a 16-element, 56-m long vertical array in approximately 100-m deep shallow water.

5:20

4pSPb5. Blind deconvolution of sources of opportunity in ocean waveguides using bilinear channel models. Ning Tian, Justin Romberg, and Karim G. Sabra (Georgia Inst. of Technol., 75 5th St. NW, Atlanta, GA 30308, ningtian@gatech.edu)

We develop a general algorithmic framework for acoustic imaging using sources of opportunity. While these sources (e.g., ships at known locations) emit high-energy signals, we do not in general have knowledge of the precise waveforms (signatures) of these signals. Consequently, both Channel Impulse Responses (CIRs) and unknown source signals need to be simultaneously estimated from only the recorded signals on a receiver array using blind deconvolution, which is generally an ill-posed problem without any a priori information or additional assumptions about the underlying structure of the CIRs. By exploiting the typical ray-like arrival-time structure of the CIRs between a surface source and the elements of a vertical line array (VLA) in ocean waveguides, we apply a principle component analysis technique to build a bilinear parametric model linking the amplitudes and arrival-times of the CIRs across all channels for a variety of admissible ocean environments. The bilinear channel representation further reduces the dimension of the channel parametric model compared to linear models. We then develop a truncated power interaction deconvolution algorithm by applying the bilinear channel model to the traditional subspace deconvolution method. Numerical and experimental results will demonstrate the robustness of this blind deconvolution methodology.

5:35

4pSPb6. Experimental demonstration of passive acoustic tracking using a library of nearby sources of opportunity. Christopher M. Verlinden, Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu), Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), William S. Hodgkiss, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Acoustic sources can be tracked in the marine environment using a method that is similar to traditional matched field processing but differs in that it uses a library of data-derived replicas in place of modeled replicas. In order to account for differing source spectra between library and target vessels, cross-correlation functions are compared instead of comparing acoustic signals directly. Measured replicas are extrapolated to fully populate a search grid using waveguide invariant theory. The method has been demonstrated experimentally for localizing surface contacts in a shallow water (150 m) environment. Further, we present experimental validation of the method in deeper water and discuss the extension of the technique to subsurface contacts

5:50

4pSPb7. Localization of circularly towed sources using machine learning methods. Emma Reeves (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu), Haiqiang Niu (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Localization of a source towed in a circular geometry is accomplished using machine learning methods, wherein the relationship between received pressure and source range is learned directly from the acoustic data. Feed-forward neural networks (FNN), sup- port vector machines (SVM), and random forests (RF) have accurately estimated the range of a source towed in a linear geometry in shallow water (arXiv: 1701.08431v3). Here, we apply FNN, SVM, and RF to a circular source tow using data from the SCEx17 experiment to examine the effect of varying source-receiver geometries. The Sample Covariance Matrix (SCM) is constructed, vectorized and used as the machine learning input. For FNN and RF, these input vectors are combined to form a d x N matrix and an additional preprocessing step is applied to improve classification results. The input matrix is projected onto a d x k basis formed from the top k eigenvectors of its scatter matrix. This compact representation results in improved computation time and performance for FNN and RF compared with using the SCM inputs.

Session 4pUW

Underwater Acoustics: Arctic Acoustics

Matthew Dzieciuch, Cochair
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Jason D. Sagers, Cochair
Environmental Science Laboratory, Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

1:00

4pUW1. The 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment (CANAPE): A preliminary report. 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), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (Dept. of Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA), Jonathan D. Nash (Dept. of Earth, Ocean & Atmos. Sci., Oregon State Univ., Corvallis, OR), and John N. Kemp (Appl. Ocean Phys. & Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Arctic Ocean is undergoing dramatic changes in both ice cover and ocean structure. The Canada Basin Acoustic Propagation Experiment (CANAPE), which includes both deep and shallow water components, was designed to understand the effects of changing Arctic conditions on low-frequency propagation and ambient noise. The deep-water component, which is reported on here, includes a yearlong experiment in the Canada Basin during 2016–2017, preceded by a short Pilot Study during July–August 2015. During 2016-2017, a Distributed Vertical Line Array (DVLA) receiver with 60 Hydrophone Modules was moored within a six-element acoustic transceiver array with a 150-km radius. Environmental measurements on the DVLA include 28 Sea-Bird MicroCATs and upward- and downward-looking Acoustic Doppler Current Profilers (ADCPs) located below the hydrophone array. The acoustic transceivers had sources at 175-m depth and 15 Hydrophone Modules located above the sources. Environmental measurements on the transceiver moorings include ice-profiling sonars, upward-looking ADCPs on the subsurface floats, and 10 temperature sensors located below the acoustic transceivers. The one-year deployment provides measurements at least partially in open water during summer, in the marginal ice zone (MIZ) as it transitions across the array during the spring and autumn, and under complete ice cover during winter.

1:15

4pUW2. Acoustic propagation from the Canadian Basin to the Chukchi Shelf. Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Ying-Tsong Lin (AOPE Dept MS 11, Woods Hole Oceanogr. Inst., Woods Hole, MA), Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), and Sean Pecknold (Defence Res. and Development Canada-Atlantic, Dartmouth, NS, Canada)

The Pacific Arctic Region, encompassing the Bering, Chukchi, Western Beaufort, and Eastern Siberian shelves and seas, has experienced decadal changes in atmospheric conditions, seasonal sea-ice coverage, and seawater temperature. In the summer of 2016, the Canada Basin Acoustic Propagation Experiment (CANAPE) was conducted to understand the changing soundscape and to explore the use of acoustic signals as a remote sensing tool in the modern Arctic. During the experiment, low-frequency signals from five tomographic sources located in the Canada Basin were recorded by a short vertical line array of hydrophones deployed from a research vessel. The recordings were made at seven stations located in the Canada Basin, on the continental rise, and on the Chukchi Shelf. The propagation distances ranged from 50 km to 500 km, and the propagation conditions changed from ducted by the Beaufort Lens in the basin to upward refracting on the continental shelf. Multiple measurements of the sound speed profile were acquired at each station to characterize the temporal and spatial variability of the sound speed field. This talk examines the range-dependent measurements and explains the observed variability in the received signals through propagation modeling. [Work sponsored by ONR.]

1:30

4pUW3. Underwater sound propagation variability over the Chukchi Sea continental slope. Ying-Tsong Lin, Weifeng G. Zhang, Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, Woods Hole, MA 02543, ytlin@whoi.edu), Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, Newark, DE), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Sean Pecknold (Defence Res. and Development Canada-Atlantic, Dartmouth, NS, Canada)

In the Canada Basin and Chukchi Sea regions, a vertical sound duct can be formed between the Pacific Summer Water Layer on the top and the Atlantic Water Layer on the bottom, providing an acoustic pathway connecting the deep basin and the shallow shelf over the Chukchi Sea continental slope. Previous studies have shown that the shelfbreak circulation (specifically upwelling), the sub-mesoscale eddies spun off the shelfbreak jet, and the ice coverage are the three major causes of the temporal and spatial variability of the Pacific Summer Water Layer in the region. In this paper, numerical simulations utilizing the Parabolic-Equation (PE) method are conducted to investigate the sound propagation variability over the Chukchi Sea shelfbreak and slope, along with an idealized ocean circulation model providing the fundamental basis of water-column fluctuations. The sound pressure sensitivity kernel derived from the acoustic propagator in the PE method is also used to provide physical insights into the sound propagation variability. The acoustic data over the northern Chukchi Sea shelfbreak collected during the 2016 shallow water cruise of the Canada Basin Acoustic Propagation Experiment will be examined and compared with the numerical simulations. [Work supported by the Office of Naval Research.]

1:45

4pUW4. Oceanographic measurements on Chukchi shelf break during fall 2016. Mohsen Badiey, Andreas Muenchow, Justin Eickmeier, and Lin Wan (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Newark, DE 19716, badiey@udel.edu)

During the Canada Basin Acoustic Propagation Experiment (CANAPE) in 2016, two extended shipboard oceanographic measurements were conducted simultaneously with the acoustic propagation from deep water to the Chukchi shelf-break region. These shipboard measurements were aimed at understanding the oceanographic variability including along the shelf eddy formation and upwelling around the shelf break region. We utilize the measured oceanographic data to construct the environmental input for acoustic models. While there was no ice formation or coverage during the observation period, the effects of upwelling and the eddy formation on acoustic propagation were present. This paper demonstrates the results measured in the experiment in the context of temporal and spatial variability of the water column in the Chukchi shelf region. [Work supported by ONR Ocean Acoustics].

2:00

4pUW5. Acoustic characterization of the new Arctic using mobile acoustic sources. John E. Joseph, D. Benjamin Reeder, and Liam J. Doyle (Oceanography, Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 315A, Monterey, CA 93943, jejoseph@nps.edu)

The Naval Postgraduate School (NPS) participated in Ice Exercise 2016 (ICEX-16), a multi-national naval exercise conducted in the Beaufort Sea during March 2016. Operating at the remote Ice Camp SARGO, NPS deployed several conductivity, temperature and depth (CTD) sensors to capture oceanographic variability to 500 m depth while performing a series of propagation tests. Four mobile mid-frequency sources transmitted signals for approximately 10 hours each to a pair of vertical line array receivers positioned in the field to investigate depth, range, angular and specular characteristics of acoustic propagation and their correlation to variability in oceanographic structure and under-ice conditions. CTD data indicated significant variability in sound speed at 50 m depth where cold, fresh mixed-layer water interfaces with contrasting warm, saline Pacific Summer Water (PSW) that lays immediately below it. The data also show a persistent and stable subsurface sound channel existed as a result of the PSW with peak temperature at 80 m situated above colder Pacific Winter Water (PWW), resulting in a sound channel axis near 140 m depth. Both features have important implications on sonar performance in the Arctic. Modeled and measured transmission loss are compared to quantify the effects.

2:15

4pUW6. Limitations of a Gaussian Beam Model for low-frequency acoustic propagation under ice. Sean Pecknold (DRDC Atlantic, PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), Diana F. McCammon (Maritime Way Sci., Waterville, NS, Canada), and Dale D. Ellis (Maritime Way Sci., Dartmouth, NS, Canada)

Ray-based propagation models are often used to simulate underwater acoustic communications signals. For longer-range low frequency acoustic communications, particularly under ice cover, it is necessary to determine if a ray-based model can give accurate results. An elastic ice reflection algorithm was added to the Bellhop Gaussian beam model, and the results for mid- and low-frequency propagation were compared to results obtained using a normal modes model adapted for an elastic ice reflection coefficient, and to the OASES wavenumber integration model. The effects of simplified boundary conditions, frequency, and ice thickness on amplitude and phase matching are examined.

2:30

4pUW7. Modeling annual variations in high-frequency scattering from sea ice. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu)

As sea ice evolves naturally through its different growth and decay structures, so too does it manifest important changes in physical properties affecting its acoustic response. Variations in the porosity, permeability, and roughness of the ice lead to changes in scattered and reflected acoustic energy as well as the degree of attenuation. Knowledge of the long-term variations of high-frequency under-ice scattering over annual formation and melting cycles is essentially non-existent as only a limited number of measurements have been performed. Owing to this lack of data, fundamental questions remain as to the dominant mechanisms influencing the under-ice scattering process at any given time. In this work, we will show modeling results for high-frequency acoustic scattering from different types of sea ice as it forms and melts, highlighting the relative influence of various ice parameters on mean scattered levels and the dependence of these levels on frequency and grazing angle. Results yield insight into controlling factors and lead to recommendations for planned long-term measurements from moored instruments.

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, 5 December

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Studio 7
Acoustical Oceanography	7:30 p.m.	Salon A/B/C
Animal Bioacoustics	7:30 p.m.	Salon F/G/H
Architectural Acoustics	7:30 p.m.	Studio 9
Musical Acoustics	7:30 p.m.	Studio 4
Physical Acoustics	7:30 p.m.	Balcony L
Psychological and Physiological Acoustics	7:30 p.m.	Balcony M
Structural Acoustics and Vibration	8:00 p.m.	Studio 7

Committees meeting on Wednesday, 6 December

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Balcony M
Signal Processing in Acoustics	7:30 p.m.	Salon D

Committees meeting on Thursday, 7 December

Committee	Start Time	Room
Noise	7:30 p.m.	Studio 2
Speech Communication	7:30 p.m.	Salon A/B/C
Underwater Acoustics	7:30 p.m.	Salon F/G/H

Session 5aAA

Architectural Acoustics: Topics in Architectural Acoustics: Measurements, Modeling, and Isolation

Ian B. Hoffman, Cochair
Peabody Institute, Johns Hopkins Univ., 1 E Mount Vernon Place, Baltimore, MD 21202

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—8:30

Contributed Papers

8:35

5aAA1. Effect of acoustic computer model precision on auralization perception. Brandon M. Westergaard and Timothy Hsu (School of Music, Georgia Inst. of Technol., 840 McMillan St. NW, Atlanta, GA 30318, brandon.westergaard@gatech.edu)

Most modern acoustic simulation programs use geometric acoustics to auralize the sound within a space, providing acceptable results in a relatively quick manner. Still, because of the limitations of geometric acoustics in computer simulations, it is of particular interest to investigate the relationship between model geometry, simulation parameters, and the listener's perception of auralizations created from iterations of an acoustic model. In the context of this study, differences between auralizations were investigated from a perceptual standpoint, using the subjective judgment of listeners. A modified version of the Multiple Stimulus test with Hidden Reference and Anchor methodology was used to define a threshold at which an increase in model precision no longer results in perceived differences between auralizations, aiming to inform efficient modeling practices in practical applications. Acoustic simulations were performed on a set of speech and music venues modeled at incremental geometrical precision regarding their real-life counterparts, ultimately resulting in a series of auralizations that reflected changes in modeling precision. Single factor analysis of variance and a series of two-sample t-tests indicate that increasing acoustic model precision may not always result in perceptually relevant differences.

8:50

5aAA2. Room impulse response synthesis with device diffraction via image source method and finite element analysis. Aidan Meacham and Andrew Unruh (Knowles Electronics, 331 Fairchild Dr., Mountain View, CA 94043, aidan.meacham@knowles.com)

A method to create an accurate reproduction of the soundfield on the surface of a device was developed through the combination of the image source method and anechoic finite element analysis simulations. Typically, the image source method is unable to model diffraction of sound pressure around a device, and it would be impractical to model the wave equation in an entire room with finite element analysis due to computational constraints. By combining the two techniques, however, individual impulse responses from a precomputed finite element dataset can be assembled like image sources to derive transfer functions from sources in space to receivers on the surface of a device. The finite element dataset is comprised of anechoic simulations of plane waves incident on the device, where the number and direction of plane waves are chosen to maximize angular resolution. The direction of arrival from a given image source is matched with a simulated plane wave, and the corresponding anechoic impulse response is delayed and attenuated according to the image source's time of travel and reflection path. Each reflection is constructed in this manner, and all are summed to create a synthesized reverberant impulse response that accurately portrays diffraction around the device.

9:05

5aAA3. Acoustic characterization of Penn State's recording studio above and below the Schroeder frequency. Andrew D. Kinzie and Daniel A. Russell (Acoustics, Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azk5643@psu.edu)

In most performance spaces, rooms are large enough that most of the audible frequency range falls above the Schroeder frequency, whereas smaller spaces have a significant audible range below the Schroeder frequency. The goal of this study was to characterize the acoustic performance of the control room and live room of Penn State's recording studio. Both rooms have important roles to play in the operation of a studio which depend on their acoustic character. COMSOL models for each room were created to understand the modal behavior below the Schroeder frequency and measured modes were visualized from frequency response measurement points in one plane at a one foot spacing. For frequencies above the Schroeder frequency, an ODEON model was created and validated against measured impulse responses, where metrics such as T60 and EDT have been considered. By comparing the computer simulations to the measurements, one is able to make recommendations for improvements and provide audio engineers with insight on how they can adapt to the room and refine their craft.

9:20

5aAA4. Assessment of a variable acoustic system proposal in a real case. Merve Esmebasi, Ali Murat Tanyer (Architecture Dept., Middle East Tech. Univ., Ankara 06800, Turkey, merveesmebasi@gmail.com), and Mehmet Caliskan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

This paper is about testing a variable acoustic system proposal under development in a real case. The variable acoustic system is designed as alternating slot panels and prototyped via using Arduino and Processing. Digital interface to control the system is designed in Processing and via Arduino board digital commands are converted to physical movements. After the possibility of construction of variable perforated panel system is simply clarified, perforated panels of the system are produced via CNC routers. Panel positions are defined regarding perforation characteristics in terms of slot width and distances between slots. Changing these parameters is linked to perforation ratio of the system, while air gap and porous material behind are kept unchanged. Sound absorption capabilities of variable perforations are examined with Impedance tube measurements. Measurement results of variable slot panels, as a promising variable acoustic system, are transferred to ODEON room acoustics software, to assess the system in terms of room acoustics. The amphitheater at METU Faculty of Architecture Building, used for different functions from lectures to musical performances, is

selected to demonstrate the capability of the proposed variable acoustics system to improve acoustical conditions in an amphitheater.

9:35

5aAA5. Experimental characterization of the Green's function in a room using sparse reconstruction principles. Efren Fernandez-Grande (Acoust. Technol., Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk) and Rasmus Ellebæk Christiansen (Mech. Eng., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Measuring the Green's function over the entire volume of a room would typically require an unfeasible number of measurements, due to requirements on spatial sampling. To alleviate the need for excessive measurements, sparse reconstruction methods can be employed, as they make it possible to reconstruct a seemingly undersampled signal. The present study proposes a method for acquiring experimentally the Green's function in a room by measuring directly the mode shapes of the room, based on the conception that any mode can be expanded into a number of propagating waves. If the modes are described in the wavenumber domain (as a plane-wave expansion), sparse reconstruction methods can be employed, under the implicit assumption that each mode shape is represented as the superposition of a small number of plane waves. In addition, it is assumed that the medium is source-free and homogeneous. The methodology is examined numerically and verified experimentally, based on measurements in a lightly damped rectangular room.

9:50

5aAA6. A concert hall database of US and European halls: Preliminary measurements and results. Matthew T. Neal and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

Subjective perception in the realm of concert hall acoustics, specifically overall impression, is a difficult problem to approach. Ideally, this type of work should be done using realistic concert hall auralizations, allowing direct comparison of a wide variety of rooms from around the world. Currently, measurements are being taken in concert halls across the United States and in Europe. Halls to be measured were selected from an online survey of researchers and consultants around the United States and Europe. Final selections were made to ensure that a wide variety of hall shapes, sizes, and reverberation times were included in the database, given the available travel resources. Measurements have been made using a 32-element spherical microphone array, a three-way omnidirectional sound source, and a directional sound source. The directional sound source is a 20-element compact array that can reconstruct the frequency-dependent radiation patterns of different orchestral instruments. An overview of the halls included in the database and measurement results will be presented to illustrate the variety within the database. As well, plans for upcoming subjective testing to piece-apart the perception of preference in concert hall acoustics, considering both the hall and individual taste, will be discussed. [Work supported by NSF Award 1302741.]

10:05–10:20 Break

10:20

5aAA7. Absorption: The misunderstandings about what can be measured and the newest ideas of how to perform these measurements. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

We will present the latest research into the measurement of absorption, the pitfalls, and some possible ways out of the these pits. We will also present new information on methods derived from the measuring of diffusion that can be applied to measuring absorption.

10:35

5aAA8. Overcoming field impact isolation performance failures due to faulty flooring installation practices. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

There is very strong demand by new owners of luxury condominiums for hard-surfaced flooring. In some cases, there is a medical need. In most cases, the reasons are personal preference. However, most high-end condominium homeowner associations (HOA) have enacted regulations requiring that upper level dwelling unit owners, making flooring upgrades from any existing flooring to hard-surfaced flooring, meet higher minimum field impact insulation class (FIIC) or impact sound rating (ISR) performance standards than is typically required by building codes. Failure to comply usually causes the HOA to require restoration to prior flooring conditions within a certain time period, with fines and potential litigation for restoration delays. This paper presents field performance test results for several renovated condominiums in different multi-family residential complexes that had successfully passed HOA impact performance tests by using customized sample underlayments and the owners' desired hard-surfaced flooring, but failed to pass the HOA impact performance requirements after final installation due to faulty installation practices, in spite of explicit verbal and written installation instructions provided to the owners and their flooring contractors. Examples of methods verifying faulty installations are discussed and successful remedies are presented, along with respective passing test results.

10:50

5aAA9. Sound isolation of large transformers mounted on raised floors. Angelo J. Campanella (Campanella Assoc., 3201 Ridgewood Dr., Hilliard, OH 43026, a.campanella@att.net)

Transformers for substation service mounted on elevated floors often cause spurious sound to be transmitted to the spaces below it. Magnetostriction vibration of the massive iron core at 120 Hz and its harmonics drives the supporting floor to emit sound into the spaces below. Contemporary vibration isolation design fails to prescribe the proper isolation method since the mass of that transformer considerably exceeds that of the participating mass of the concrete floor. A vibration isolation paradigm is proposed of a two-body system in inertial space; the participating mass of the concrete floor being the second active body; the virtual mass of the pair along with the spring stiffness determines the resonance frequency. The ratio of that resonant frequency to that to be isolated is a measure of the expected isolation. For good isolation, the resonant frequency should be 1/10th that of the vibration to be isolated. The floor participating mass is approximated as a square one-half wavelength of the thin concrete slab bending wave. Recent case history supporting this model is described.

11:05

5aAA10. Archaeoacoustics: Testing and evaluation methodology. Vincent C. Paladino (PO Box 29 Lodi, Lodi, NJ 07644, bioenginearing@gmail.com)

Testing and measurement procedures of acoustic properties in caves, caverns, and buildings have been conducted along with EEG measurements in order to determine the effects of acoustic resonance on the human brain. The results have given rise to questions and speculative models concerning the role of acoustic phenomena in the development of human culture. Conceptualizations of the role of sound in sociocultural evolution add needed dimensions to archaeology, and challenge the science of acoustics. The problem of methodology needs to be addressed, and standards must be established. The relationship of testing techniques and the relevance of measurement parameters to the spaces and material culture found in these locations is central to the successful establishment of archaeoacoustics as a valid scientific study.

11:20

5aAA11. VampireVerb: A surreal simulation of the acoustics of Dracula's Castle. Alex Chechile, Constantin Basica, Elliot K. Canfield-Dafilou, and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, chechile@ccrma.stanford.edu)

This work introduces VampireVerb, a parameterized modal reverberator created from acoustics measurements recorded at Bran Castle in Transylvania, Romania, popularly called "Dracula's Castle." With limited evening access to the space, balloon pop sources were recorded using an A-format microphone and handheld recorder in several acoustically interesting rooms, including the armory. Spatial impulse responses were estimated from the balloon pop recordings, and a modal reverberator was designed to simulate the measured acoustics. In the main rooms, reverberation times T30 in the range [0.7 s, 1.2 s] were noted. A long, narrow stairwell and the torture chamber were acoustically dry, and did not produce the sound one would typically associate with Dracula's Castle. To bridge this gap between the actual and imagined acoustics, the mode frequencies and dampings were parametrized so as to interactively vary the size of the space and materials present, transforming the recorded rooms into large, stone spaces with the long reverberation times one would expect in Dracula's Castle. Finally, a nonlinear process was embedded in the reverberator architecture, producing a reverberant noise process modulated according to the evolving input signal spectral envelope, thus making VampireVerb both a traditional reverberator, and an instrument for compositional and performance applications.

FRIDAY MORNING, 8 DECEMBER 2017 — SALON A/B/C, 9:00 A.M. TO 10:15 A.M.

Session 5aAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Carolyn M. Binder, Cochair
Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

Boris Katsnelson, Cochair
Marine Geosciences, University of Haifa, Mt. Carmel, Haifa 31905, Israel

Contributed Papers

9:00

5aAO1. Confounding effects of environment-dependent propagation on automated classification of cetaceans. Carolyn M. Binder (Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca) and Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada)

Significant effort has been made over the last few decades to develop automated passive acoustic monitoring (PAM) systems capable of classifying cetaceans at the species level. The utility of such systems depends on their ability to operate across a wide range of ocean acoustic environments; however, anecdotal evidence suggests that site-specific acoustic propagation characteristics impact the performance of PAM systems. This is because properties of the ocean acoustic environment can be markedly different between regions and seasons in which PAM is used to observe cetaceans. Variability in propagation characteristics leads to differences in how each cetacean vocalization is distorted as it propagates along the source-receiver path. Unless these differences are accounted for, the acoustic environment will bias estimates of a PAM system's performance. The impact of environment-dependent propagation on an aural classifier was assessed using a pulse propagation model. Simulations were conducted that showed the sensitivity of classifier performance to sound speed profile type. Furthermore, it was found that classifier performance was range-dependent, largely due to changes in signal-to-noise ratio (SNR); however, in some environments only 60% of the performance reduction was attributed to decreasing SNR, indicating that signal distortion—resulting from environment-dependent propagation—had a large impact on performance.

9:15

5aAO2. Calculating a sensitivity kernel for the final cutoff of received energy in a broadband channel impulse response. Matthew Dzieciuch and Bruce Cornuelle (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu)

The final cutoff of a deep-water ocean acoustic tomography reception is often interpreted through a modal analysis. An alternative approach is presented here. First, a stable travel-time is generated by using an estimator-correlator to account for the statistical variability of the final low-angle arrival with the assumption of an appropriate coherent bandwidth and depth coherence. A time-series of these travel-times can then be tracked using the Viterbi algorithm. Then, the resolvable, trackable peaks are interpreted using the travel-time sensitivity kernel (TSK). The TSK has been previously used to account for the full-wave effects of ray-like arrivals. But the TSK machinery can be adapted to any processing strategy with the application of the chain-rule. The question of how to interpret an arrival produced by an Estimator-Correlator vs. an arrival produced by standard processing is examined. The feasibility of this approach is first analytically determined with the Pekeris waveguide and then an example from a recent ocean acoustics experiment is presented.

9:30

5aAO3. Whispering gallery waves in horizontal plane in vicinity of curved isobaths in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il) and Pavel S. Petrov (V.I.Il'ichev Pacific Oceanological Inst., Vladivostok, Russian Federation)

Interference structure of the sound field in horizontal plane in shallow water with parameters variable in horizontal plane is studied within the framework of the so-called 3D problem. Formally, this problem can be solved using separation of the vertical coordinate (depth) in the wave equation in supposition that depth dependence of the sound field is described by adiabatic waveguide modes. Remaining part is two-dimensional dispersive wave equation which can be solved by different methods (PE, ray approximation, and modal decomposition). It is shown in the paper that in a shallow water waveguide with variable bathymetry with curvilinear isobaths (Laguna, lake) there exist specific solutions of this equation, concentrated in horizontal plane approximately along isobath lines (whispering gallery waves), up to formation of the waveguide modes in the horizontal plane. Number of modes and their shape depend on position of the source, frequency and radius of curvature. Remark that this whispering gallery modes can exist in real conditions, for example, for frequency about a few hundreds Hz and radius of curvature about 5–10 km. Analytical expression and results of modeling using ray approximation and PE are presented for real shallow water conditions. [Work was supported by ISF, grant 565/15.]

9:45

5aAO4. Data-driven decomposition of long-term echosounder time series from ocean observatories. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Valentina Staneva, Bernease Herman (eScience Inst., Univ. of Washington, Seattle, WA), and Aleksandr Aravkin (Dept. of Appl. Mathematics, Univ. of Washington, Seattle, WA)

Recent advances in technology have produced a deluge of acoustic data that offer opportunities to study ecological processes at scales that are not possible previously. A prominent example is the continuous data flow from the fleet of upward-looking echosounders installed by the Ocean Observatories Initiatives (OOI) at diverse global locations. However, it is unclear if conventional echosounder analysis routines are effective in analyzing these data sets, due to the generally scarce ground-truth resources and limited empirical knowledge at many locations. In this study, we explored the use of signal decomposition methods in discovering daily patterns of marine organism activities in long-term echosounder time series. Using non-negative matrix factorization (NMF), we show that the echograms can be decomposed into a weighted combination of discrete components, each with acoustic features that can be exploited for inferring the underlying biological assemblage. In addition, the component weights provide an avenue to capture echogram variations in a significantly reduced dimensional space, which can be utilized to describe changes in the ecosystem. Building on these results, we are developing new formulations to incorporate continuity in both time and space to construct a fully scalable framework for data-driven discovery using long-term echosounder data.

10:00

5aAO5. Cross frequency cross mode coherences using transport theory. Sivaselvi P, Tarun K. Chandrayadula (Ocean Eng. Dept., IIT Madras, Chennai, Tamil Nadu 600036, India, oe15d008@smail.iitm.ac.in), and John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

Internal waves induced acoustic scattering cause mode coupling. Previous studies on mode scattering effects used narrowband approaches to predict mode energies and time coherences. However, these single-frequency predictions do not explain statistics such as time-wander and multipath spread. In order to address this issue, this paper develops a broadband scattering model that uses the frequency coherences of the modes. For the frequency coherence predictions, this paper uses the physics based transport theory approach. The predictions are setup for an environment similar to the Philippine Sea deep water experiment, which used frequencies around 250 Hz, bandwidth of 100 Hz, and propagation ranges from ~150 km to ~450 km. The physics based predictions are compared with complementary coupled mode based Monte-Carlo simulations. The coherences are used to predict two types of statistics. The first models the time spread for each respective mode. The second uses the cross-frequency cross-mode coherences to predict the stability of the ray time-fronts. The model uses up to mode 75 to predict the multipath time spread in the finale region and a few preceding ray-like arrivals. For the finale region, this paper uses the first 20 modes and the preceding rays 75 modes.

FRIDAY MORNING, 8 DECEMBER 2017 BALCONY M, 9:00 A.M. TO 11:45 A.M.

Session 5aBA

Biomedical Acoustics: Therapeutic Ultrasound

Douglas Miller, Chair
Radiology, Univ. Michigan, 3240A Medical Science I, 1301 Katherine Street, Ann Arbor, MI 48109-5667

Contributed Papers

9:00

5aBA1. Ultrasonic cavitation-enabled treatment for therapy of hypertrophic cardiomyopathy: Proof-of-principle. Douglas Miller, Xiaofang Lu, Chunyan Dou, Yiying I. Zhu, Mario L. Fabiilli (Radiology, Univ. Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Gabe E. Owens (Pediatrics Cardiology, Univ. of Michigan, Ann Arbor, MI), and Oliver D. Kripfgans (Radiology, Univ. Michigan, Ann Arbor, MI)

Ultrasound myocardial cavitation enabled treatment (MCET) creates scattered microlesions in the myocardium, which can be accumulated to produce a desired macrolesion. MCET was applied to the SS-16^{BN} rat model of hypertrophic cardiomyopathy (HCM) for proof-of-principle as a means for myocardial reduction. A focused ultrasound transducer was targeted using 10 MHz imaging (10S, GE Vivid 7) to the left ventricular wall of anesthetized rats in a warmed water bath. Pulse bursts of 4 MPa peak rarefactional pressure amplitude were intermittently triggered 1:8 heartbeats during 10 min infusion of a microbubble suspension. Methylprednisolone was given to reduce initial inflammation and Losartan was given to improve healing. MCET significantly reduced the targeted wall thickness ($n = 11$) at 28 d post treatment by 16.2% ($P<0.01$) relative to shams ($n = 8$), with myocardial strain rate and endocardial border displacement reduced by 34% and 29%, respectively. This demonstrates sufficient effect for a therapeutic outcome similar to surgical myectomy or alcohol ablation. Premature ECG complexes and plasma troponin measurements at the time of treatment were found to be useful to gauge optimal and suboptimal treatments, and thus aid in achieving a desired impact. With clinical translation, MCET therapy should fill the need for a new non-invasive HCM therapy option.

9:15

5aBA2. An efficient boundary element solver for trans-abdominal high-intensity focused ultrasound treatment planning. Pierre Gélat, Seyyed Reza Haqshenas (UCL Mech. Eng., London, United Kingdom), Timo Betcke (UCL Dept. of Mathematics, London, United Kingdom), Elwin van 't Wout (Pontificia Universidad Católica de Chile, Santiago de Chile, Chile), and Nader Saffari (UCL Mech. Eng., London, United Kingdom, n.saffari@ucl.ac.uk)

High-intensity focused ultrasound (HIFU) is a promising treatment modality for the non-invasive ablation of pathological tissue in many organs, including the liver. Since many patients are not suitable candidates for liver surgery, the possibility to locally deposit thermal energy in a non-invasive way would bear significant clinical impact. Optimal treatment planning strategies based on high-performance computing numerical methods are expected to form a vital component of a successful clinical outcome in which healthy tissue is preserved and optimal focusing achieved, thus compensating for soft tissue heterogeneity and the presence of ribs. The boundary element method (BEM) is an effective approach for this purpose because only the boundaries of the ribs and soft tissue regions require discretization, as opposed to standard approaches which require the entire volume around the ribcage to be meshed. A Galerkin discretized Burton-Miller formulation used in combination with preconditioning and matrix compression techniques was applied to compute the acoustic field generated by a focused array transducer in the presence of a layer of abdominal fat, a human ribcage model and liver tissue. The results demonstrate the effectiveness of this dedicated BEM algorithm for trans-abdominal HIFU treatment planning.

9:30

5aBA3. Pulse inversion therapy for improved monitoring of blood-brain barrier opening. Antonios Pouliopoulos, Mark Burgess, and Elisa Konofagou (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, Columbia University Medical Campus, New York City, NY 10032, a.pouliopoulos@columbia.edu)

Microbubble-based ultrasound therapy has enabled non-invasive and reversible opening of the blood-brain barrier (BBB). However, the skull limits our ability to monitor microbubble activity due to high attenuation and beam aberrations. In ultrasound imaging, pulse inversion is used to cancel echoes from linear scatterers by summing the signal obtained from consecutive positive (phase: 0^o degrees) and negative (phase: 180^o degrees) pulses, thus facilitating imaging of non-linear scatterers such as microbubbles. Here, we adapt the pulse inversion technique to improve monitoring of BBB opening, by transmitting consecutive therapeutic pulses of inverse polarity. Pulse inversion therapy (PIT) was achieved by synchronizing the emission of inverted short pulses (pulse length: 2–3 cycles, PRF: 2 kHz, and pressure: 400 kPa) through a focused 0.5 MHz therapeutic transducer driven by two function generators. A concentric P12-5 linear array was used to passively capture the microbubble emissions. Absolute time-of-flight information was introduced in the beamforming since emission and reception were synchronous. PIT suppressed the signals from linear scatterers within the focal region by up to 6 dB in a gelatine phantom containing microbubbles and in mice *in vivo*, compared to positive-only pulses. Ongoing *in vivo* work aims at correlating the BBB opening with the microbubble signal identified by PIT.

9:45

5aBA4. Investigation of molecular mechanisms induced by a combination of high-intensity focused ultrasound and chemical anti-cancer agents. Heng Yu, Hakm Y. Murad, Daishen Luo (Biomedical Eng., Tulane Univ., 6823 St. Charles Ave., 440 Lindy Boggs Bldg., New Orleans, LA 70118, hyu3@tulane.edu), Andrew Sholl (Dept. of Pathol., TulaneUniv., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Tumor exposure to a combination of chemical treatment and physical forces causes chemical modifications and structural changes in cancer cells which inhibit their ability to survive and proliferate. High-Intensity focused ultrasound (HIFU), a unique non-invasive and non-ionizing physical force modality, has great potential as an adjuvant for chemical anti-cancer therapy. *In vitro* and *in vivo* studies conducted in our laboratory showed that HIFU synergistically enhanced tumor destruction by ethanol injection or chemotherapeutic drugs and reduce the potential for tumor metastasis. The objective of this study was to investigate the molecular mechanisms behind adjuvant anti-cancer effects of HIFU. To do this, we employed a combination of flow cytometry and western-blot analysis as well as *in vitro* and *in vivo* assays of tumor growth and cell adhesion. We showed HIFU drastically reduced the metastatic potential of chemically treated cancer cells via overproduction of heat-shock protein 70 (HSP70), death receptor Fas, its ligand FasL and TNF-α receptor. Although Hsp70 plays a key role in cancer initiation and progression, its overproduction induced by HIFU interferes with NF-κB signaling, thereby causing apoptosis and reduced expression of adhesion molecules required for metastasis. All these factors lead to phenotypic changes in surviving cancer cells that reduce their aggressiveness.

10:00

5aBA5. Ultrasound-enhanced molecular therapy for axon neurogenesis. Asis Lopez, Adrian Jones, Bridget Daugherty, and Damir Khismatullin (Biomedical Engineering/Bioinnovation, Tulane Univ., 6823 St. Charles Ave., New Orleans, LA 70118, alopez12@tulane.edu)

An estimated 55M individuals experience spinal cord injuries (SCIs), 230K in the United States. When a nerve is injured, the environment prevents healing of neurons and myelin. This is due to the presence of myelin inhibitors, growth factor not re-expressing, and glial tissue scarring rapidly. In order to bypass this mechanism nerves need to regenerate quickly, and unfortunately, no effective method currently exists to stimulate neurogenesis to the central nervous system. We propose an innovative non-invasive method to stimulate neurogenesis through low-intensity focused or unfocused ultrasound irradiation. We developed an *in vitro* system integrating unfocused ultrasound (UFUS) with neuron axon dynamics using a 3D microenvironment for neurogenesis. Dorsal root ganglion (DRG) neurons are used for our studies and microsurgically removed at day 15 and E18 Sprague Dawley cortical neurons from BrainBits, LLC. To determine if stimulation of axon growth occurs, we measure density and distance using phase contrast imaging and perform Scholl Method. In parallel, we began an *in vivo* protocol exposing the spinal cord of mice and using two-photon microscopy. We tested and demonstrated a variety of operating parameters to identify optimal conditions that stimulate axon DRGs by UFUS.

10:15–10:30 Break

10:30

5aBA6. *In vivo* investigation of high-intensity focused ultrasound combined with thermally triggered chemotherapy for liver cancer treatment. Gray Halliburton, Hakm Y. Murad, Monica Kala (Biomedical Eng., Tulane Univ., 31 Mcalister Dr., New Orleans, LA 70118, ghallibu@tulane.edu), Yueheng Zhang (Chemical and Biomolecular Eng., Tulane Univ., New Orleans, LA), Andrew Sholl (Pathol. and Lab. Medicine, Tulane Univ. School of Medicine, New Orleans, LA), Vijay John (Chemical and Biomolecular Eng., Tulane Univ., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

High-Intensity Focused Ultrasound (HIFU) induces rapid tissue heating and mechanical disruption and also provides noninvasive release of chemotherapy from temperature-sensitive-liposomes (TSLs). We evaluated the combination of Sorafenib-loaded TSLs (SfTSLs) and HIFU via analysis of cell viability, tumor growth, and long-term survival. Liposomes were encapsulated with Sorafenib at 10μM with a glass transition temperature of 43°C. Hep3B human liver cancer cells were placed in PCR tubes to mimic dense tumor aggregates. Aggregates were treated with HIFU alone, SfTSLs alone, or SfTSLs + HIFU. HIFU exposure was applied for 30 seconds at acoustic output powers of 8.7 and 12W. Cell viability and proliferation were measured over 4 days post-treatment via AnnexinV/PI and WST-8 staining. Xenografted tumors were created via injection of 1.0×10^6 Hep3B cells within Matrigel into flanks of athymic-nude mice. Tumors at 8–10 mm were separated into the following groups: Control (sham), HIFU, SfTSLs, and SfTSLs + HIFU. Tumors were measured daily with endpoints at 5 days, 14 days, or long-term survival. Dissected tumors were H&E stained and evaluated by a blinded pathologist. The *in vitro* data indicates that Hep3B cells exposed to SfTSLs + HIFU have a significantly lower viability and proliferation rate. Similarly, *in vivo* results show SfTSLs + HIFU reduces tumor growth and increases survival.

10:45

5aBA7. Focused-ultrasound mediated anti-alpha-synuclein antibody delivery for the treatment of Parkinson's disease. Hairong Zhang, Carlos S. Sanchez, Nancy Kwon (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, hz2440@columbia.edu), Vernice R. Jackson-Lewis, Serge Przedborski (Dept. of Neurology, Columbia Univ., New York, NY), and Elisa Konofagou (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

Parkinson's disease (PD) is associated with the selective death of dopaminergic (DA) neurons in the substantia nigra pars compacta (SNpc). While the specific cause of the neuronal loss remains elusive, the abnormal accumulation of alpha synuclein (α-syn), a major constituent of Lewy bodies, is considered to play a central role in the pathology of PD. Previous studies have shown the potential of immunotherapy with antibodies against α-syn, but such treatments remain ineffective due to the presence of the blood-brain barrier (BBB), which hinders most therapeutic agents to diffuse to the brain parenchyma. Therefore, in this study, we used focused ultrasound (FUS) in conjunction with microbubbles to transiently and noninvasively open the BBB and deliver anti-α-syn monoclonal antibodies (mAb) to the brains of transgenic PD mouse models. Preliminary histological findings demonstrate that the FUS promotes the delivery of anti-α-syn mAb to the transgenic mice overexpressing the human A53T or A30P α-synuclein. We hypothesize that weekly FUS treatments with mAb will help ameliorate histopathological deficits such as alpha-synuclein and Lewy bodies represented in these mouse models of PD. Ongoing work aims to explore the potential of FUS-mediated drug delivery to achieve both neuroprotection and neurorestoration for the treatment of Parkinson's disease.

11:00

5aBA8. Variation of high intensity therapeutic ultrasound pressure field characterization: Effects of hydrophone choice, nonlinearity, spatial averaging, and complex deconvolution. Yunbo Liu, Keith A. Wear (FDA, 10903 New Hampshire Ave., WO62RM2126, Silver Spring, MD 20993, yunbo.liu@fda.hhs.gov), and Gerald Harris (FDA, Rockville, MD)

Reliable acoustic characterization is fundamental for patient safety and clinical efficacy during high intensity therapeutic ultrasound (HITU) treatment. Technical challenges, such as measurement variation and signal analysis, still exist for HITU exposimetry using ultrasound hydrophones. In this work, four hydrophones were compared for pressure measurement: a robust needle hydrophone, a small PVDF capsule hydrophone, and two different fiber-optic hydrophones. The focal waveform and beam distribution of a single element HITU transducer (1.05 MHz and 3.3 MHz) were evaluated. Complex deconvolution between the hydrophone voltage signal and frequency-dependent complex sensitivity was performed to obtain pressure waveforms. Compressional pressure (p_+), rarefactional pressure (p_-), and focal beam distribution were compared up to 10.6/−6.0 MPa (p_+/p_-) (1.05 MHz) and 20.65/−7.20 MPa (3.3 MHz). The effects of spatial averaging, local nonlinear distortion, complex deconvolution, and hydrophone damage thresholds were investigated. This study showed a variation of at least 10-15% between different hydrophones during HITU pressure characterization.

11:15

5aBA9. Development of a treatment simulation platform—Application to probe optimization and tissues characterization. Raphaël Loyet (Univ. Lyon, Université Lyon 1, INSERM, LabTau, 151 Cours Albert Thomas, Lyon 69424, France, raphael.loyet@inserm.fr), Sylvain Chatillon (CEA, LIST, Gif sur yvette, France), Françoise Chavrier (Univ. Lyon, Université Lyon 1, INSERM, LabTau, Lyon, France), loic sifferlen (CEA, LIST, Gif sur yvette, France), Ayache Bouakaz (INSERM U930, Tours, France), Stéphane Leberre, and Pierre Calmon (CEA, LIST, Gif sur yvette, France)

In order to ease the development and optimization of new transducers and therapeutic protocols, a unified tool to easily simulate 3D pressure fields created by HIFU in human tissues has been developed jointly by INSERM and CEA-LIST. This includes three propagation algorithms: 1/ a GPU implementation of Rayleigh integral for fast linear propagation in human soft tissues, 2/ a dynamic ray tracing algorithm for the simulation of propagation in heterogeneous or inhomogeneous media (i.e., bones, skull, heated tissues, etc.), 3/ a Westervelt solver for non-linear propagation. These algorithms have been validated and implemented in a unified graphical user interface and can be easily interchanged to compare the results or to target a specific research topic. A finite volume difference solver for Bio Heat Transfer Equation (BHTE) has also been implemented. It can be used with all the HIFU propagation algorithms. It is used to simulate thermal effects of HIFUs and thermal dose in tissues and phantoms. Parametric studies targeting probe optimization and better characterization of tissues will be presented. [Work supported by French Nation Research Agency (ANR SATURN -15-CE19-0016).]

11:30

5aBA10. High-intensity focused ultrasound is synergistic with anti-neoplastic drugs that target endoplasmic reticulum stress. Hakm Y. Murad, Emma P. Bortz (Biomedical Eng., Tulane Univ., 6823 St. Charles, New Orleans, LA 70118, hmurad@tulane.edu), Partha Chandra, Debasis Mondal (Pharmacology, Tulane Univ., New Orleans, LA), and Damir Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Previous studies from our laboratory show that combination of High-Intensity Focused Ultrasound (HIFU) and anti-neoplastic agents not only reduces cellular viability and proliferation, but also shifts cellular populations to late apoptotic/necrotic stages. Nelfinavir, a well-tolerated HIV protease-inhibitor, is known to increase endoplasmic reticulum (ER) stress, and is thus being repurposed as an anti-neoplastic agent. This led us to combine Nelfinavir with HIFU in order to create a safe and more effective treatment for patients with prostate cancer. 2.7 million cells/ml suspensions of DU145 and C4-2B prostate cancer cells were placed in PCR tubes and split into 4 treatment groups: Control, HIFU alone, Nelfinavir(2μM) alone, and Nelfinavir(2μM) + HIFU. HIFU treatment was applied for 30 seconds at an acoustic output power of 8.7 W. Apoptosis and necrosis of cells was measured via AnnexinV/PI staining, cell proliferation was measured by a WST-8 assay, and ROS was stained using cellular ROS detection assay kit. The population percentage of Late Apoptotic/Necrotic cells and ROS production was found to be significantly greater, while proliferation rate was significantly reduced post exposure to Nelfinavir + HIFU 24 and 72h post treatment. HIFU and Nelfinavir have a synergistic effect on both increasing cellular death via apoptosis/necrosis and decreasing proliferation *in vitro*.

Session 5aPA

Physical Acoustics, Structural Acoustics and Vibration: Nonlinear Elasticity in Geomaterials

Marcel Remillieux, Cochair
Geophysics Group (EES-17), Los Alamos National Laboratory, Mail Stop D446, Los Alamos, NM 87545

Pierre-yves Le Bas, Cochair
Geophysics Group (EES-17), Los Alamos National Laboratory, Mail Stop D446, Los Alamos, NM 87545

Invited Paper

8:00

5aPA1. Stress corrosion crack imaging in stainless steel using the time reversed elastic nonlinearity diagnostic and nonlinear resonant ultrasound spectroscopy. Brian E. Anderson, Sarah M. Young, Stephen M. Hogg, and Joshua F. Gregg (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu)

Spent nuclear fuel is often stored in stainless steel canisters in the United States. Stainless steel is susceptible to Stress Corrosion Cracking (SCC). This presentation will discuss progress on the use of the Time Reversed Elastic Nonlinearity Diagnostic (TREND) and Nonlinear Resonant Ultrasound Spectroscopy (NRUS) to determine whether SCC is present and attempt to quantify the depth of the cracking. NRUS is the measurement of the amplitude dependence of a sample's resonance frequency, which occurs because of a softening of the elastic modulus in damaged media. NRUS provides a global indication of damage in a sample. TREND employs time reversal acoustics, which focuses wave energy at various points of interest to excite localized high amplitude. The amplitude dependence of this localized energy allows pointwise inspection of a sample. [This work was funded by the Nuclear Energy University Program of the U.S. Department of Energy through a subcontract from Los Alamos National Laboratory.]

Contributed Papers

8:20

5aPA2. Nonuniform electromechanical efficiency and point defects in lithium niobate. Chandrima Chatterjee and Igor Ostrovskii (Dept. of Phys. and Astronomy, Univ. of Mississippi, 108 Lewis Hall, PO Box 1848, University, MS 38677-1848, cchatter@go.olemiss.edu)

The electromechanical efficiency (EME) and distribution of point defects (DPD) are measured from $LiNbO_3$ samples at room temperature. The EME is estimated by reading an electric potential (V) generated while applying a mechanical force to a local point. The DPD is measured by Photoluminescence (PL) taken from samples. PL is excited by 310-nm light and optical spectra are registered from 350 to 900 nm. The defects are identified by their characteristic peaks. The applied mechanical force and intensity of 310-nm light are kept constant while scanning V and PL along Z-axis or normal to Z. The F-center and Fe^+ have been shown to be sensitive to local electrical polarization. Thus, they may influence EME in a local point of crystal. The experimental data from bulk and wafer samples show a nonuniform distribution of the F-center and Fe^+ defects. For example, a 1.9-mm-thick Y-cut wafer demonstrates periodicity in defect concentration of 1 to 2 mm along the Z-axis. Similar periodicity is detected in the potential V along the same crystallographic axis. The revealed correlation between defects and EME along with their nonuniform distribution may be a physical basis of nonlinear phenomena involving piezoelectricity including nonlinear elasticity, acoustical memory, etc.

8:35

5aPA3. Three-dimensional description of nonlinear elasticity in rocks: Modeling and experiments. Marcel Remillieux (Los Alamos National Lab., Geophys. Group (EES-17), Mail Stop D446, Los Alamos, NM 87545, mcr1@lanl.gov), Martin Lott, Vincent Garnier (LMA, CNRS, UPR 7051, Aix-Marseille Université, Centrale Marseille, Marseille, France), Pierre-yves Le Bas, Timothy J. Ulrich (Los Alamos National Lab., Los Alamos, NM), and Cedric Payan (LMA, CNRS, UPR 7051, Aix-Marseille Université, Centrale Marseille, Marseille, France)

We study theoretically and experimentally the mechanisms of nonlinear and nonequilibrium dynamics in geomaterials through dynamic acousto-elasticity testing. In the proposed theoretical formulation, the classical theory of nonlinear elasticity is extended to include the effects of conditioning. This formulation is adapted to the context of dynamic acousto-elasticity testing in which a low-frequency "pump"-wave induces a strain field in the sample and modulates the propagation of a high-frequency "probe"-wave. Experiments are conducted to validate the formulation in a long thin bar of Berea sandstone. Several configurations of the pump and probe are examined—the pump successively consists of the first longitudinal and first torsional mode of vibration of the sample while the probe is successively based on (pressure) P- and (shear) S-waves. The theoretical predictions reproduce many features of the elastic response observed experimentally, in particular, the coupling between nonlinear and nonequilibrium dynamics and the three-dimensional effects resulting from the tensorial nature of elasticity. [This work was supported by the French National Research Agency through the ENDE program (Grant No. ANR-11 RSNR 0009) and the U.S. Department of Energy through the Fossil Energy program (Grant No. FE-634-15-FY15).]

8:50

5aPA4. Nonlinear shear wave resonator consisting of a relaxing material. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Soft materials such as rubbers, polymers, and tissue exhibit low shear wave speeds, facilitating the generation of shear waves with large acoustic Mach numbers. In addition to finite-amplitude effects that result from cubic nonlinearity, plane shear wave propagation in these materials is subject to frequency-dependent attenuation and dispersion that result from viscoelastic effects. A wave equation for plane shear waves in a relaxing material is obtained from a nonlinear Zener constitutive model that accounts for cubic nonlinearity as well as the attenuation and dispersion associated with relaxation. The wave equation is used to analyze a one-dimensional shear wave resonator comprised of a nonlinear relaxing material that is shaken at one end and free at the other. For excitation of the lowest mode the wave equation is approximated by an augmented Duffing equation, and the resulting frequency-response equation is compared with numerical finite-difference solutions of the original wave equation. Frequency responses are presented as functions of both drive amplitude and relaxation time. The model predicts behavior resembling that reported by Andreev *et al.* [*Acoust. Phys.* **57**, 779 (2011)] for experiments on a similar resonator comprised of plastisol and a head mass. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

Invited Papers

9:05

5aPA5. Experimental measurements and bristle friction modeling of nonlinear hysteresis loops and harmonic generation in rock fractures. Seth Saltiel, Brian Bonner (Earth and Environmental Sci., Lawrence Berkeley National Lab, 1 Cyclotron Rd., Berkeley, CA 94720, ssaltiel@lbl.gov), Tushar Mittal, Brent Delbridge (Earth and Planetary Sci., UC Berkeley, Berkeley, CA), and Jonathan Ajo-Franklin (Earth and Enivironmental Sci., Lawrence Berkeley National Lab, Berkeley, CA)

Frictional properties affect the propagation of high-amplitude seismic waves across rock fractures. Laboratory evidence suggests that these properties can be measured in active seismic surveys, potentially offering a route to characterizing friction *in situ*. We present experimental results from a subresonance torsional modulus and attenuation apparatus that utilizes micron-scale sinusoidal oscillations to probe the nonlinear stress-strain relation at a range of strain amplitudes and rates. Nonlinear effects are further quantified using harmonic distortion; however, time series data best illuminate underlying physical processes. The low-frequency stress-strain hysteretic loops show stiffening at the sinusoid's static ends, but stiffening is reduced above a threshold frequency. This shape is determined by harmonic generation in the strain; the stress signal has no harmonics, confirming that the fractured sample is the source of the nonlinearity. These qualitative observations suggest the presence of rate-dependent friction and are consistent between fractures in three different rock types. We propose that static friction at the low strain rate part of the cycle, when given sufficient "healing" time at low oscillation frequencies, causes this stiffening cusp shape in the hysteresis loop. While rate-and-state friction is commonly used to represent dynamic friction, it cannot capture static friction or negative slip velocities. So, we implement a bristle friction model, which describes this process and produces similar results.

9:25

5aPA6. Nonlinear acoustics evaluation of CO_2 exposed sandstone. James Bittner (Univ. of Illinois, 205 N. Mathews St. MC-250, Urbana, IL 61801, jbittn2@illinois.edu), Pierre-yves Le Bas (Los Alamos National Lab., Los Alamos, NM), John Popovics (Univ. of Illinois, Urbana, IL), and Paul A. Johnson (Los Alamos National Lab., Los Alamos, NM)

In an effort to better understand the hygro-thermo-mechanical behavior of geologic CO_2 reservoir material, we investigate the nonlinear behavior of elastic wave propagation in Berea sandstone samples, which are used as a standard for reservoir rock formations. Nonlinear characterization methods, including resonant ultrasound spectroscopy (RUS), dynamic acousto-elasticity (DAET), and single-impact nonlinear resonance techniques, are applied to pristine, damaged (distributed microfractures), and CO_2 injected Berea samples; conventional linear vibrational and wave propagation measurements are also applied to the samples. The results of these sensitive test methods are compared to reveal the characteristics of geologic reservoir materials that are most affected by varying microstructural and environmental conditions. An analysis of the work also leads to potential bases for test methods that could be deployed in the field in the future to monitor the condition of reservoir formations and lead to better understanding of CO_2 injection-induced seismic events. This work is done within the framework of the $GSCO_2$ center for geologic storage of CO_2 from the U.S. department of Energy whose purpose is to better understand CO_2 sequestration to make it safer and more efficient. As such the results obtained by elastic waves measurement will also be compared to other testing, providing an insight on the physical origin of the nonlinear behavior of geomaterials.

9:45

5aPA7. Dynamic acousto-elastic testing with an ultrasonic tomography probe: Application to fractured rock characterization. Jacques Riviere and Philippe Roux (ISTerre, Inst. of Earth Sci., Grenoble Alpes Univ., ISTerre Université Grenoble Alpes CS 40700, Grenoble Cedex 9 38058, France, jacques.riviere@univ-grenoble-alpes.fr)

We study nonlinear elastic phenomena at the laboratory scale to help interpret the subtle velocity changes observed in the Earth's crust, for instance, during strong ground motion, earthquake slip processes or Earth tides. Dynamic Acousto-Elastic Testing (DAET) provides unprecedented details on the nonlinear elastic response of consolidated granular media (e.g., rocks, concrete), including tension/compression asymmetry, hysteretic behaviors as well as conditioning and relaxation effects. Such technique uses a pump-probe scheme where a high frequency, low amplitude wave probes the state of a sample that is dynamically disturbed by a low frequency, large amplitude pump wave. While previous work typically involved a single pair of probing transducers, here we use two dense arrays of ultrasonic transducers to image a sample of Westerly granite with a complex fracture. We apply double beamforming to disentangle complex arrivals and conduct ray-based and finite-frequency tomography using both travel time and amplitude information. By comparing images obtained before, during, and after the pump wave disturbance, we are able to locate and characterize nonlinear sources within the sample. We discuss their locations with regard to low velocity/high attenuation zones and relate our observations to large-scale data.

10:05–10:25 Break

Contributed Paper

10:25

5aPA8. Flaw detection of wellbore systems by combining time reversal methods and nonlinear acoustic measurements. Carly M. Donahue and Pierre-yves Le Bas (Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, cmd@lanl.gov)

A wellbore system consisting of a steel pipe cased in cement is susceptible to cracking and delamination. Such flaws greatly reduce the integrity of the wellbore and can lead to significant leakage. To resolve the location and extent of the defects, we have combined a unique time reversal method with nonlinear acoustic evaluation. The time reversal method generates a compact high-amplitude wave only in a localized area that can be used to probe the acoustic properties spot by spot of a three dimensional region. Additionally, the nonlinear acoustic signal an object exhibits is far more sensitive to damage and defects than conventional linear analysis. The nonlinear behavior can be locally determined by observing the changes in elastic properties as the amplitude of the time-reversed focus wave is increased. The experimental setup consist of an array of piezoelectric transducers for generating the focused energy and a laser vibrometer mounted on a scanning frame for focusing and measuring nonlinearity on the interior of the steel tubing.

Invited Papers

10:40

5aPA9. Nonlinear softening of unconsolidated granular earth materials. Charles Lieou (Los Alamos National Lab., MS D446, Los Alamos, NM 87545, clieou@lanl.gov), Eric Daub (Ctr. for Earthquake Res. and Information, Univ. of Memphis, Memphis, TN), Robert Guyer (Physics, Univ. of Nevada Reno, Reno, NV), and Paul A. Johnson (Los Alamos National Lab., Los Alamos, NM)

Unconsolidated granular earth materials exhibit softening behavior due to external perturbations such as seismic waves, namely, the wave speed and elastic modulus decrease upon increasing the strain amplitude. In this letter, we describe a theoretical model for such behavior. The model is based on the idea that shear transformation zones (STZs)—clusters of grains that are loose and susceptible to contact changes and rearrangement—are responsible for plastic deformation and softening of the material. We apply the theory to experiments on simulated fault gouge composed of glass beads, and demonstrate that the theory predicts nonlinear resonance shifts and reduction of the P-wave modulus, in agreement with experiments. The theory thus offers insights on the nature of the critical state prior to failure on earthquake faults.

11:00

5aPA10. The role of nonlinear ultrasound in the diagnosis of early-stage damage in heterogeneous materials. Gun Kim (Carle Illinois College of Medicine, The Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., MC-251, Urbana, IL 61801-2325, mcguncall@gmail.com), Tae Sup Yun (School of Civil and Environ. Eng., Yonsei Univ., Seoul, South Korea), Jin-Yeon Kim, Kimberly Kurtis, and Laurence Jacobs (School of Civil and Environ. Eng. / G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Predictable noninvasive evaluation of engineering materials requires a more reliable sensing technique capable of providing quantitative information of early stage damage. Nonlinear ultrasound (NLU) is a promising candidate because it provides a direct measure of the nonlinear elastic behavior of materials. NLU excels in the direction and quantification of damage that originates at or beneath the material's microscale. This talk will present a procedure for the second harmonic generation (SHG) measurements using nonlinear Rayleigh surface waves. This technique quantifies material nonlinearity through the acoustic nonlinearity parameter, β. Specifically, microscale material characterization of physical/chemical phenomena in heterogeneous materials will be reviewed by means of the acoustic nonlinearity parameter, β. The results reveal how the SHG technique can provide the quantitative relationship between the acoustic nonlinearity parameter and the damage state of these materials. Last, new strategies for the application of the SHG technique will be discussed with an emphasis on bio-engineering materials and rocks.

Contributed Papers

11:20

5aPA11. The soil plate oscillator: Nonlinear mesoscopic elasticity of layers of glass beads in flexural vibration. Emily V. Santos and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com)

The soil plate oscillator (SPO) apparatus consists of two circular flanges sandwiching and clamping a thin circular elastic plate. In this presentation, uniform spherical glass beads—representing a nonlinear mesoscopic elastic material—are supported at the bottom by the plate and stiff cylindrical sidewalls of the upper flange. A small magnetic disk centered and fastened below the plate is driven by an AC coil placed below the magnet. Nonlinear tuning curves of the magnet's acceleration are measured by driving the coil with a swept sinusoidal signal applied to a constant current amplifier. Experiments are first performed to measure the peak resonant frequency vs. mass loading as glass beads are added to make a level column above the plate. Experiments are repeated using 1,2,3,…,10 mm diameter beads to demonstrate how the frequency first decreases and then eventually increases for each bead diameter. Next, ten (separate) tuning curve experiments are performed using a fixed column of 350 grams of beads for each diameter. The backbone curves (peak acceleration vs. corresponding resonant frequency) exhibit a linear region with comparable slopes. While the detailed curvature vs. bead diameter reveals more structure, a bilinear hysteresis model fits the results.

11:35

5aPA12. Broadband acoustic wave experiments in borehole configurations. Huajun Fan (College of Geophys. and Information Eng., China Univ. of Petroleum (Beijing), 18 Fuxue Rd., Changping, Beijing 102249, China, hj.fan@hotmail.com) and David Smeulders (Dept. of Mech. Eng., Eindhoven Univ. of Technol., Eindhoven, Netherlands)

A borehole in the ground may penetrate rocks which are porous, permeable, and fractured. Pressure transients in the borehole will therefore cause viscous fluid to flow into and out of the wall of the borehole. This forced flow consumes some energy and affects the phase velocity of the waves

traveling along the fluid column. These two effects can be evaluated to extract information about the rocks adjacent to the borehole, which is of paramount importance for the oil and gas industry. This work discusses laboratory wave experiments in a 7.5 m long vertical shock tube. Rock samples containing a vertical borehole and pre-fabricated fracture configurations are installed in the test section and filled with water. The shock tube generates broadband pressure transients in the borehole. These are measured in the borehole at variable depth by means of a sliding pressure probe. In this way borehole microseismograms are constructed from repetitive wave experiments. Borehole wave modes are identified using coherence analysis. It is found that the presence of fractures significantly influences the transmission and reflection of borehole waves over the fracture zone. Moreover, there is a strong correlation between fracture aperture and borehole acoustic properties.

FRIDAY MORNING, 8 DECEMBER 2017 STUDIOS FOYER, 9:00 A.M. TO 12:00 NOON

Session 5aSC

Speech Communication: Bilingual and Non-Native Speech Perception and Production (Poster Session)

Charlotte Vaughn, Chair
Linguistics, University of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290

All posters will be on display from 9:00 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 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

5aSC1. Context effects on accentedness ratings. Charlotte Vaughn and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, cvaughn@uoregon.edu)

Intelligibility and accentedness are largely independent judgments (Derwing & Munro, 1995); a speaker rated as highly accented can be quite intelligible. The context of listeners' exposure to accented speakers has been shown to affect adaptation to speakers in terms of intelligibility (e.g., Tzeng *et al.*, 2016), suggesting that implicit comparison between examples facilitates learning of systematic properties of accented speech. However, it is unknown whether context affects accentedness ratings, which are often assumed to be stable properties of speakers. To better understand the susceptibility of accent ratings to context effects, the present study examines listeners' accentedness judgments of native and non-native speech taken from the ALLSSTAR Corpus (Bradlow *et al.*, 2010). A target set of the same accented speech samples was embedded in a variety of task contexts, varying whether the stimuli were randomized among speakers or blocked by speaker, or presented at the beginning or end of the experiment. Results will shed light on how stable accentedness judgments are, and begin to test the extent to which these judgments are tied to particular items, speakers, accents, or listeners. Determining whether accentedness ratings are affected by context is a step toward understanding what factors listeners use when making accentedness ratings.

5aSC2. The perception of English Vowels by Korean Native Speakers: Concerning the influence of the manner of articulation for consonants following vowels. Heesun Han (Graduate School of Lang. and Culture, Osaka Univ., 1-8 Machikaneyama, Toyonaka, Osaka 560-0043, Japan, kenkyuhhs@gmail.com) and Takeshi Nozawa (Graduate School of Lang. and Culture, Osaka Univ., Kusatsu, Japan)

The purpose of this study is to explore the perception of English vowels by Korean native speakers. It focuses on difficulty levels of perception for English vowels in relation to the sound environment. The sound environment is determined by differences in the manner of articulation of consonants following vowels. The experiment consists of 24 test words formed with six American English vowels(/i, ɪ, ɛ, æ, ɑ, ʌ/) embedded in four different sound environments (/hVt/, /pVt/, /pVn/, /pVl/). The experiment compares two words paired for each sound environment, examining judgment accuracy for each combination and the influence of sound environments. Twenty Korean native speakers in their 20s or 30s participated in the experiment. The results found that participants could rarely differentiate /i/ and /ɪ/, and /ɛ /and /æ/ in all sound environments. Moreover, judgment accuracy was lower overall for the /pVn/ and /pVl/ environments. For /pVl/, in particular, Korean native speakers misjudged /ʌ/ as /ɑ/, confirming that difficulty levels differed according to the sound environments.

5aSC3. Second language English learners' perception of foreign-directed speech. Kathrin Rothermich (Speech, Lang. and Hearing, Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269-1085, kathrin.rothermich@uconn.edu), Kristin Mello, Emily Turco, Larissa Lemes, Erika Fernandez, and Susan Bobb (Psychology, Gordon College, Wenham, MA)

Previous research has shown that people adapt the way they speak depending on the perceived comprehension level of the listener (Uther *et al.*, 2007) including when they speak to foreigners. While the acoustic properties of accommodations such as foreign speech are well documented, few studies have investigated how second language learners in particular interpret them. The present study investigated English language learners' (ELLs) perception of different speech types. Participants heard four types of auditory stimuli (casual, clear, infant-directed, and foreign-directed) spoken by four different speakers (two males, two females) and evaluated the extent to which the speaker was easy to understand, competent, condescending, friendly, and respectful. Perceptual ratings showed an interaction between speech type and question type ($p < 0.001$). Casual speech was least intelligible, least competent, least friendly, and least respectful compared to all other speech types. No effects were found for condescension. The implications of these results suggest that ELLs perceive speech accommodation of any type as positive.

5aSC4. Non-native perception of Vietnamese tone contrasts. Madeleine Oakley (Linguist, Georgetown Univ., 3700 O St. NW, Washington, DC 20057, mo643@georgetown.edu)

A listener's ability to discriminate non-native sound contrasts has been shown to be largely influenced by the listener's native phonological system (Best *et al.*, 2003; Tyler *et al.*, 2014; Tskuda, 2012). Looking specifically at

suprasegmental contrasts, experimental results suggest that the degree to which pitch is used to distinguish lexical items in a speaker's L1 influences how well they are able to detect non-native tone contrasts (Schaefer & Darcy, 2014). However, these results are based only on Thai tones. The present study shows that native speakers of English (a stress language), and Mandarin (a tone language) do not perform significantly differently in their ability to perceive Vietnamese tone contrasts. Results from an ABX categorization task show that native speakers of English are not more likely to make errors in categorizing Vietnamese tones than native speakers of Mandarin, and both groups have difficulty perceiving the difference between the low falling tone and the low falling-rising tone. These results suggest that the acoustic properties of a tone, such as the register and the contour, contributes more to how well non-native speakers can discriminate a contrast than does the L1 of the listener.

5aSC5. Training new second language category formation. Anna M. Schmidt and Nora Hassan (Speech Pathol. & Aud., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

The Speech Learning Model (Flege, 1995) hypothesizes that, in second language (L2) learning, the greater the difference the learner perceives between in a sound from a first language (L1) category and a sound from an L2 category, the more likely it is that a new L2 category will be formed. Finding an L1 category that is not similar in some way to an L2 category is difficult. However, a L2 sound, that is similar to an L1 sound, could be consciously taught as a new allophone of that sound in a way that focuses attention on phonetic differences. In this study, Chinese L1 speakers, from a region of China where /n/ and /l/ in word initial position in their dialect (and in Mandarin) are produced as /n/, will be taught a dark Arabic /l/ as a new position specific allophone to be used in word initial position in English and Mandarin Chinese. Results from perception and production of /l/ and /n/ before and after training will be reported.

5aSC6. Phonological neighborhood density and speech production in L1 and L2 English speakers. Veera S. Vasandani (Univ. of Minnesota, Minneapolis, MN), Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR), Jennifer S. Kim, and Benjamin Munson (Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Phonological Neighborhood Density (PND) affects speech production latencies, word durations, and acoustic-phonetic detail (e.g., Baese-Berk & Goldrick, 2009; Buz & Jaeger, 2016; Fox, Reilly, & Blumstein, 2015; Munson, 2013; Munson & Solomon, 2004; Wright, 2004). The nature of these effects has been debated actively. Some have argued that these effects reflect intentional articulatory modifications to increase intelligibility, while others have suggested that they are the consequence of planning processes. The purpose of this presentation is to examine whether these effects are present in second-language speakers of English whose first language is Korean. Because the mechanisms underlying these effects are unclear, examining whether non-native speakers also show sensitivity to PND will inform our understanding of the mechanisms that drive them. Moreover, studying non-native speakers of varying proficiency allows us to infer whether PND-driven variation in production emerges over the course of second-language acquisition. The production of multiple repetitions of high- and low-PND CVC words was collected from 17 native adult speakers of English and 19 adult L2 speakers of English whose L1 was Korean. Estimates of vocabulary size were also collected for both groups. Analyses of production latencies and vowel acoustics are ongoing. [Funding provided by a University of Minnesota Multicultural Summer Research Opportunity Program award to the first author.]

5aSC7. Effects of clear speech and language background on multimodal perception of English fricatives. Sylvia Cho (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca), Allard Jongman (Linguist, The Univ. of Kansas, Lawrence, KS), Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), and Joan A. Sereno (Linguist, The Univ. of Kansas, Lawrence, KS)

Research shows that acoustic modifications in clearly enunciated fricative consonants (relative to the plain, conversational productions) facilitate auditory fricative perception. However, clear-speech effects on visual fricative perception have received less attention. A comparison of auditory and visual (facial) clear-fricative perception is particularly interesting since sibilant fricatives in English are more auditorily salient while non-sibilants are more visually salient. This study thus examines clear-speech effects on audio-visual perception of English sibilant and non-sibilant fricatives. Native English perceivers and non-native perceivers with different L1 fricative inventories (Mandarin, Korean) identified clear and plain fricative-vowel syllables in audio-only (AO), visual-only (VO), and audio-visual (AV) modes. The results across perceiver groups and speech styles showed an overall visual benefit and auditory dominance (AV>AO>VO). Comparisons of styles revealed clear-speech benefits in AO across fricatives and groups, but different patterns were noted in the visual conditions. In VO and AV, clear speech helped the more visually salient non-sibilant identification for native perceivers; however, clear-speech benefits were less prominent in non-natives' perception of the non-sibilants, which are non-existent in their L1s. These findings are discussed in terms of the relative audio-visual weighting that benefits perception in clear speech as a function of input saliency and perceiver experience.

5aSC8. L2 duration characteristics of Mandarin tones by Japanese learners. Yue Sun (Waseda Univ., Room 702, 3 Chome 14-9, Okubo, Shinjuku-ku, Tokyo 169-0072, Japan, yue.cherry.sun@gmail.com), Jinsong Zhang (Beijing Lang. and Culture Univ., Beijing, China), and Yoshinori Sagisaka (Waseda Univ., Tokyo, Japan)

In order to get a better understanding of the acquisition of Mandarin Chinese prosody by L2 learners, we investigate the temporal aspects of Mandarin tones in running speech for native Chinese speakers and Japanese L2 learners. Significant differences are found between natives and learners when generating tones with varying syllable durations depending on their position in the utterance as well as on specific syntax constraints associated with them. Among the tone types, the shortening of tone 3 pronouns at the beginning of utterances and the lengthening of tone 4 syllables at the end of utterances remain difficult for learners to generate, even at a higher overall proficiency level. Throughout the duration analysis, we could speculate the existence of stress control in Chinese syllable timing.

5aSC9. A comparative acoustic analysis of Korean vowels by native and non-native speakers. Na-Young Ryu (Linguist, Univ. of Toronto, 30 Charles St. West, Unit 903, Toronto, ON M4Y1R5, Canada, nayoung.ryu@mail.utoronto.ca)

The primary purpose of this study is to examine L1 transfer in L2 vowel acquisition (Flege 1995,1996) by comparing L1 vowels with L2 Korean vowels produced by Korean native speakers and both Mandarin and English learners of Korean. It would predict which L2 Korean vowels are relatively difficulty or easy to produce for the non-native speakers based on acoustic similarities and dissimilarities between their L1 vowels and L2 Korean vowels. A total of 68 female speakers participated in a word-list reading task. For acoustic analysis, the formant frequencies (F1 and F2) and vowel duration were measured. Results demonstrated that there were cross-language differences in both vowel quality and duration. Both Mandarin and English learners of Korean perform well when producing Korean [a, i] vowels, but have difficulty producing Korean vowel contrasts [ʌ]-[o], [o]-[u], [ɨ]-[u]. In terms of vowel duration, Korean vowels were the shortest, English vowels were the longest, and Mandarin vowels were intermediate between the two. Overall, the L2 Korean vowel duration of both Mandarin and English speakers was too short, compared to their L1 vowel productions, to have a Korean native-like performance of vowels.

5aSC10. Effect of frequency shifts on talker recognition in native and foreign-accented speech. Michelle R. Kapolowicz, Daniel R. Guest, Vahid Montazeri, and Peter F. Assmann (Behavioral and Brain Sci., The Univ. of Texas at Dallas, 800 West Campbell GR 4.1, Richardson, TX 75080, mrk092020@utdallas.edu)

The present study examined the effect of frequency shifts on perceived talker recognition in foreign-accented speech compared to native-accented speech. Sentences were processed using the STRAIGHT vocoder. The

spectral envelope and the fundamental frequency were shifted up or down in seven steps (3 up, 3 down plus unshifted) using scale factors of 8% and 30%, respectively, at each step. Listeners heard pairs of sentences and were asked to judge whether the identity of the talker was the same or different. Frequency shifts had similar effects for native- and foreign-accent conditions, in that listeners perceived the shifted versions as different talkers when, in fact, the talkers were the same. However, listeners were more likely to judge native-accented sentence pairs as the same talker regardless of whether or not they were the same; foreign-accented sentence pairs were more likely to be heard as different talkers. Overall, these results indicate that patterns of frequency-shifted foreign-accented speech are similar to previously reported patterns for frequency-shifted native speech; however, the small differences in patterns between the accent conditions might be attributed to listeners being less familiar with non-native speech patterns.

5aSC11. Towards automated detection of similarities and differences in bilingual speakers. Marisha Speights (Commun. Disord., Auburn Univ., 1199 Haley Ctr., Auburn, AL 36849, mls0096@auburn.edu), Noah H. Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Joel MacAuslan (Speech Technol. & Appl. Res., Bedford, MA), Rachel Blades, Donaldson Maya, Kara Swanson, Sarah Tuohy, JoHannah Ungruhe, and Karla Washington (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Bilingualism is increasingly the norm in the United States with at least 20% of Americans being bilingual. This change in linguistic demography has created a new challenge to accurately diagnose speech disorders among bilingual children. This study explores Acoustic Landmark Detection (ALD) system as an objective approach to characterizing similarities and differences in speech production in child speakers of Standard English and Jamaican Creole (JC). Eight JC-English bilingual children were recorded speaking eleven words three times in each language. Words were transcribed and entered into PROPH+ to provide: (1) Phonological Mean Length of Utterance [pMLU], (2) phonotactic structure, and (3) Percent Consonants Correct (PCC). Landmarks were hand-marked to determine probable landmark sequences based on canonical word production. Canonical landmark sequences were aligned with detected landmark sequences in both languages using the Needleman-Wunsch global alignment algorithm. Analysis revealed that if PCC indicates JC and English words are different, the mismatch between landmark tends to be lower (higher production accuracy); if phonotactics indicate JC and English words are different, mismatch tends to be slightly higher (lower accuracy); and if pMLU indicates word differences, mismatch tends to be higher (lower accuracy). Our finding support traditional linguistic expectations regarding bilingual speakers' similarities and differences.

5aSC12. Acoustic correlates of accentedness and comprehensibility in L2 Spanish judgments: A mixed-effects modeling approach. Ziwei Zhou and Charles Nagle (Iowa State Univ., 2603 Kent Ave. # 209, Ames, IA 50010, ziweizh@iastate.edu)

Two constructs—comprehensibility and accentedness—figure themselves prominently in listeners' judgments of L2 (second language) speech. Correlational analyses have shown that they make separate contributions to such judgments (Derwing and Munro, 1995, 1997, 2005). As it is important to set realistic goals for adult L2 learners by prioritizing understanding over nativelikeness (Levis, 2005), recent studies began to examine the relative contribution of linguistic aspects, especially acoustic features, to comprehensibility and accentedness (Munro & Derwing, 1999; Kang, Rubin, & Pickering, 2010; Trofimovich & Issac, 2012). Consistent with this agenda, this study first examines the reliability of scores (produced by the 9-point Lickert scale) under modern measurement framework (e.g. multi-facet Rasch measurement model). Then, the relationship between acoustic features, including the confidence values (for each utterance) obtained using Google Cloud Platform speech recognition engine as well as suprasegmental features extracted using Prosogram (Mertens, 2014), and ratings are established using mixed-effects modeling techniques. Results indicate that, even though it can be shown that comprehensibility and accentedness are statistically distinct constructs, both ratings share a common set of acoustic correlates as significant predictors—speech time, standard deviation of pitch values, and normalized pairwise variability index (nPVI).

5aSC13. Audio-visual perception of mandarin tone in clear speech. Yuyu Zeng (Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, yzengae@ku.edu), Keith Leung, Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman, and Joan A. Sereno (Linguist, Univ. of Kansas, Lawrence, KS)

Clearly enunciated speech (relative to conversational, plain speech) involves articulatory and acoustic modifications that have been shown to enhance segmental intelligibility. However, little research has explored clear-speech effects on the perception of suprasegmental properties such as lexical tone, particularly involving visual (facial) perception. Moreover, research has not determined if visual tonal cues are linguistically significant, presumably because tone production does not rely on vocal tract configurations. Investigating clear-speech effects may help validate the contribution of visual information, since improved visual tone intelligibility in clear speech would indicate the linguistic relevance of visual cues. The present study tested this hypothesis by examining the intelligibility of clear and plain Mandarin tones by native (Mandarin) and non-native (English) perceivers with auditory/visual input modalities (AO: audio-only; VO: visual-only; and AV: audio-visual). Results showed small but significant clear-speech benefits in each modality for both natives and non-natives. However, while the natives revealed an overall visual gain and auditory dominance (AV>AO>VO), visual gain appeared to be less prominent in non-native perception (AV<AO>VO in plain, AV=AO>VO in clear). These results demonstrate clear-speech facilitation in visual as well as auditory perception, particularly with native perceivers, suggesting the existence of linguistically relevant visual tonal cues.

5aSC14. Neutralization of Taiwanese tone sandhi: An acoustic study. Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Rm. 701, West Main Guanghua Bldg., Number 220, Handan Rd., Yangpu District, Shanghai, China, whouselefthand@gmail.com) and Allard Jongman (Linguist, The Univ. of Kansas, Lawrence, KS)

Taiwanese tonal alternation is realized in a circular chain shift fashion for both smooth and checked syllables. Debate regarding the processes of Taiwanese tonal alternation has centered on whether a surface tone is derived from an underlying tone, or whether a surface tone is selected without undergoing any derivation. The current study investigates this controversial issue by examining Taiwanese checked tone and smooth tone neutralization in production. In particular, we analyzed whether smooth citation and sandhi tone 55 (T51→T55) are completely neutralized in F0 contour, F0 height, and duration. We further extended the Taiwanese neutralization literature by also comparing smooth citation and sandhi tone 21 (T33→T21), checked citation and sandhi tone 53 (CT21→CT53), and checked citation and sandhi tone 21 (CT53→CT21). Non-sandhi exceptions were also included to evaluate the effect of position-in-word on F0 height and duration given that citation tones always appear in phrase-final position. Complete neutralization would indicate a surface-tone-storage-and-access point of view, whereas incomplete neutralization would suggest a derivational account for the production of Taiwanese tonal alternation. Results showed complete neutralization for checked tones after factoring out the positional effect. Results for smooth tones and theoretical implications will be also be discussed.

Session 5aSP

Signal Processing in Acoustics: Topics in Acoustic Signal Processing

Juliette W. Ioup, Chair
Dept. of Physics, Univ. of New Orleans, New Orleans, LA 70148

Contributed Papers

8:00

5aSP1. Direction of arrival estimation for conformal arrays on real-world impulsive acoustic signals. Emily Gorman (Elec. and Comput. Eng., Mississippi State Univ., 406 Hardy Rd., MS State, MS 39762, eeg87@msstate.edu), Steven L. Bunkley (USACE ERDC, Vicksburg, MS), John Ball (Elec. and Comput. Eng., Mississippi State Univ., MS State, MS), and Anton Netchaev (USACE ERDC, Vicksburg, MS)

Current methods for direction of arrival (DOA) estimation are disproportionately represented in the literature by microphone array geometry and sound source properties. A wide array of implemented methods and publications are available for uniformly spaced 2D arrays such as uniform linear arrays (ULA), uniform circular arrays (UCA), and uniform rectangular arrays (URA). Further, implemented DOA estimators are specifically designed for narrowband, continuous signals. Methods applicable to wideband signals on arbitrarily-shaped arrays are limited; alternative approaches that partition the array into sub arrays expand the number of applicable methods. For a realistic military application of a single impulse localization on a conformal microphone array, methods must be able to estimate the DOA of wideband, static, acoustic sources. DOA estimator methods' performances, capabilities, and limitations are explored on various real-world sound sources and configurations of a five-microphone conformal array.

8:15

5aSP2. Direction of arrival estimation for broadband array based on conjugate gradient. Hai X. Sun, Dandan Hai, Yongchun Miao, and Huiming Guo (School of Information Sci. and Eng. .Xiamen Univ., Xiamen, FuJian 361005, China, hxsun@xmu.edu.cn)

In Broadband array, the direction of arrival (DOA) has high computing capacity, high complexity and real time difference. To solve this issue, we have proposed fast subspace estimation based on conjugate gradient (CG) method, which used to find orthogonal vectors and span the signal subspace. Initially, training signals are not required and also avoids eigenvalue decomposition. In this paper, subspace method based on CG is combined with DOA estimation of wideband signal coherent subspace, which replaces the focus matrix and the eigenvalue decomposition of narrowband DOA estimation. Herein, we equated the computational complexity is highly reduced and also better for real-time implementation of the DOA estimation. As performance compared with old methods experimental outcomes show our method has high efficiency and low complexity. Key words: direction of arrival, conjugate gradient, subspace estimation, focus matrix, eigenvalue, etc.

8:30

5aSP3. Off-the-grid direction-of-arrival estimation with multiple measurement vectors. Yongsung Park (Seoul National Univ., Gwanak-gu, Gwanak-ro 1, Seoul National University Bldg. 36 - Rm. 212, Seoul 08826, South Korea, ysparkwin@snu.ac.kr), Youngmin Choo (Sejong Univ., Seoul, South Korea), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Acoustic signals with different direction-of-arrivals (DOAs) arrives at an array of a limited number of sensors. Compressive beamforming technique [Xenaki *et al.*, *J. Acoust. Soc. Am.* **136**(1), 260–271 (2014)] provides high-resolution results, but the conventional compressive sensing methods cause estimation degradation when basis mismatch occurs. To overcome the basis mismatch, the off the grid DOA estimation technique [Xenaki and Gerstoft, *J. Acoust. Soc. Am.* **137**(4), 1923–1935 (2015)] is suggested, but the formulation of the scheme is limited to the case of a single snapshot. In the scenario where the DOAs of the sources are stationary across multiple snapshots, the signal processing technique, which can treat the multiple snapshot data jointly, provides a more stable estimate. We extend the single snapshot grid-free DOA estimation scheme to handle the case of multiple snapshots. The multiple snapshot off the grid DOA estimation technique is demonstrated with the experimental data and this technique shows robust performance for noisy environment.

8:45

5aSP4. Eigenanalysis-based adaptive interference suppression for underwater target estimation. Suiling Ren, Lianghao Guo, and Yanli Chen (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring, Beijing 100190, China, rsl@mail.ioa.ac.cn)

It is generally difficult for a passive sonar system to localize a weak source due to the complexity and variability of the ocean environment, especially in the presence of strong interferences. In this paper, an eigenanalysis-based adaptive interference suppression method (EAAIS) is presented for estimating the weak source bearing. Assuming the target of interest (TOI) is in a known bearing sector, we first define a contribution ratio for each eigenvector of the cross-spectral density matrix (CSDM). The eigenvectors not dominated by the TOI are adaptively identified and removed for interference suppression. The remaining eigenvectors are then used to reconstruct the CSDM for TOI bearing estimation with adaptive beamforming methods. In comparison with the basic ECA method, the proposed method has better interference rejection capability and wider fields of application. Simulation and experimental results also show that the proposed method can achieve accurate localization estimates even in the presence of strong interferences.

9:00

5aSP5. Microphone arrays for efficient communications during long-duration space missions. Peter Achi and Andi Petculescu (Dept. of Physics, Univ. of Louisiana at Lafayette, Lafayette, LA 70504, pya0168@louisiana.edu)

The astronauts' ability to communicate easily among themselves or with the ship's computer should be a high priority for the success of the mission. Long-duration space habitats—whether spaceships or surface bases—will likely be larger than present-day Earth-to-orbit/Moon transfer ships. Hence an efficient approach would be to free the crew members from the relative burden of having to wear headsets throughout the spacecraft. This can be achieved by placing microphone arrays in all crew-accessible parts of the habitat. Processing algorithms would first localize the speaker and then perform speech enhancement. The background ''noise'' in a spacecraft is typically fan and duct noise (hum, drone), valve opening/closing (click, hiss), pumps, etc. We simulate such interfering sources by a number of loudspeakers broadcasting various sounds: real ISS sounds, a continuous radio stream, a poem read by one author, etc. To test the concept, we use a linear

30-microphone array driven by a zero-latency professional audio interface. Speaker localization is obtained by time-domain processing. To enhance the speech-to-noise ratio, a frequency-domain minimum-variance approach is used. Time-permitting, we will discuss array weight sensitivity to parameters such as frame length/overlap, windowing, (sub-)array structure, etc. [This work was supported by the Louisiana Space Consortium (LaSPACE).]

9:15

5aSP6. An improved robust adaptive beamforming based on worst-case performance optimization. Chao Yan, Weiyu Zhang, Peng Xu, and Mingyang Zhou (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, yanchao@mail.ioa.ac.cn)

Generally, the adaptive beamformer has better spatial resolution and much better interference rejection capability than the conventional data-independent beamformer. But, in practice, the performance of the traditional adaptive beamformer will degrade greatly if some assumptions or information on the propagation model, array parameters, and signal model are imprecise. In this paper, a new approach to robust adaptive beamforming based on optimization of worst-case performance by parameter auto-adjust (PAWCO) is proposed, which can estimate the power of the signal using Newton-Raphson interactions. The method can confirm the upper bound of the mismatch vector neither overly nor under estimated. Theoretical analysis and simulation results are presented to show the proposed beamformer can estimate the power of signal precisely while achieving better spatial resolution. The robustness of the PAWCO algorithm to environment mismatch is also demonstrated by the acoustical experimental data.

9:30

5aSP7. An inexpensive acoustic data acquisition system using a single-board microcomputer. Andrew B. Wright (Systems Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave., EIT 522, Little Rock, AR 72204, abwright@ualr.edu)

Low-cost, single-board microcomputers have become increasingly available, driven primarily by the smart phone and tablet markets. The beaglebone black (BBK, circa $50) contains a host processor, which is normally configured to run the linux operating system, and two programmable real-time units (PRU) that can access all peripherals on the device. The host processor is used to coordinate all of the human interface tasks, such as ethernet, usb, i2c, and hdmi. The PRU microcontrollers can sample either the 12-bit, 8 channel internal analog to digital converters (ADC), an external ADC, or a USB sound card, at high, precisely controlled sample rate. In this project, a BBK was programmed to sample an inexpensive external microphone through the internal ADC at 40 kHz. The power supply, external electronics, and BBK were housed in a custom-designed, 3D printed case. A unidirectional shroud for the microphone was designed and printed.

9:45–10:00 Break

10:00

5aSP8. Building air-infiltration quantification based on sound transmission loss calculated using nearfield acoustic holography. Kanthasamy Chelliah and Ralph T. Muehleisen (Argonne National Lab., 9700 S Cass Ave., Bldg. 362, Lemont, IL 60439, kchelliah@anl.gov)

This talk will demonstrate the abilities of nearfield acoustic holography (NAH) to detect and quantify leakages in building envelopes. A tonal sound source was placed inside a building model which has known leakages and microphone array measurements were obtained from the outside. Equivalent sources model based NAH was applied on the measured data to reconstruct the sound pressure field on the wall of the building model. Present results show that the NAH method was able to successfully locate the major leakages. A single microphone was used to measure the sound pressure level inside the building model which was used as a reference for quantification calculations. The difference between the inner and outer sound pressure levels was related to the area of the leakage. Various sizes of pinholes and rectangular cracks were investigated, and the detection limits of the current method were explored.

10:15

5aSP9. Representation of the combined concurrent acoustic sources in the electrical domain. Azhar Abdulaziz and Veton Këpuska (Elec. and Comput. Eng., Florida Inst. of Technol., 220 E University Blvd. Apt#1504, Melbourne, FL 32901, aabdulaziz2013@my.fit.edu)

In the electrical domain, adding samples of individual sound sources is used to represent a combined acoustic signal formed by them. It is commonly believed that, when many concurrent acoustic sources are sampled by a single microphone, it will produce samples that are equal to the linear addition of the electrical samples of those sources. The key problem about this argument is it assumes linearity between the total sound pressure level (SPL) and each individual SPL. It also assumes that the microphone voltage varies linearly with the SPL applied on it. However, it is more likely that both of those relationships are non linear, at least in theory. We believe that this is the first study that correlates the mathematical relationship between the individual acoustic signals and their combination in both electrical and acoustic domains. A theoretical new formula is derived throughout this work showing that the acoustic source samples are added in a non-linear manner. In the electrical domain, the voltages that represent each component of a multiple acoustic sources, are added together as if they are perpendicular vectors. Experimentally, adding electrical samples using the new addition formula is more accurate than the common linear one.

10:30

5aSP10. Influence of microphone position error on generating spatial null sensitivity point by line microphone array. Akio Ando (Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 3190 Gofuku, Toyama 930-8555, Japan, andio@eng.u-toyama.ac.jp)

We developed a method that picked up a sound of a target source located beyond the noise source [1]. It used an end-fire line microphone array whose direction was toward the noise source and created a spatial null sensitivity point at the noise source position. This method relied on the assumption that the precise distance between microphones was known so that all microphone outputs from the noise source became in phase by using the delay calculated by the distance. In this study, we analyzed the influence of the error of microphone distance on the performance of the method. As a result, if the real distance was larger than that of the system's knowledge, the dip of sensitivity at the noise position was separated along the direction normal to that of the line of the microphones. On the other hand, it the distance was smaller than that of the system's knowledge, the dip did not separate and just became shallower. Based on these results, we will propose how to compensate the microphone position error. [1] A. Ando, *et al.*, 3pSP3, Acoustics '17 Boston, 2017.

10:45

5aSP11. Analysis of a prediction error method employing orthonormal basis functions in adaptive feedback cancellation for hearing aids. Sahar Hashemgeloogerdi and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, 72G ClintwooDC Apts, Rochester, NY 14620, shashemg@ece.rochester.edu)

Hearing aids often suffer from acoustic feedback, which limits the achievable amplification and may severely degrade sound quality by creating howling artifacts. A potential method of feedback cancellation comprises predicting the feedback path (FP) using an adaptive filter. However, a large model error, or bias, is introduced due to signal correlations. To reduce the bias in the FP estimate, a prediction-error-method (PEM) has been used, which is based on a closed-loop identification of the FP and the auto-regressive modeling of the desired signal. This approach, which represents the FP using an FIR filter, requires a large number of parameters. Furthermore, in a reverberant environment, even a high order of the FIR filter may be insufficient to fully represent the FP, which reduces the convergence rate and limits the maximum stable gain of the system. In this contribution, we introduce a PEM-based method that utilizes orthonormal basis functions to precisely predict the acoustic FP in hearing aids. Simulation results with measured data show that the proposed method outperforms the standard PEM adaptation algorithm in terms of convergence rate and maximum stable gain.

11:00

5aSP12. Applications of operator theory and symmetry arguments to sound field recording and reproduction. Jorge A. Trevino Lopez, Shuichi Sakamoto, and Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp)

The problem of recording and reproducing sound fields using microphone/loudspeaker arrays is treated by exploiting its symmetries. This approach leads to well-known results, such as high-order Ambisonics corresponding to the requirement of the error field to be invariant under rotations. More importantly, it allows us to treat some difficult problems critical to the application of sound field recording and reproduction to virtual reality, where one or more users can freely move around a realistic environment. In this presentation, we consider three of these problems. First, we present a method to extrapolate extended sound field information from conventional spherical microphone array recordings by imposing symmetry constrains based on a priori knowledge of the sound source positions. Second, we apply the same symmetry constrains to the problem of sound field reproduction, allowing us to formulate a method of presenting spatial sound to moving listeners. Finally, we consider the problem of describing and reproducing the sound field due to a moving sound source.

11:15

5aSP13. Synthesis of non-uniformly spaced circular antenna arrays using a data-driven probabilistic model. Nicholas Misiunas (ECE, Univ. of Massachusetts, Lowell, 125 Heath St., Tewksbury, MA 01854, nmisiuna@purdue.edu), Lejun Hu, Kavitha Chandra, and Charles Thompson (ECE, Univ. of Massachusetts, Lowell, Lowell, MA)

Beamforming from non-uniformly spaced elements on a circular array of fixed radius that is steered in a two-dimensional plane is investigated. Extending previous results for randomly spaced elements on a linear array, this work examines the design of a probabilistic model using element position vectors derived through a meta-heuristic optimization process. The circular array presents a benefit in that an optimal positional distribution derived for one scan angle may be applied to other scan angles through a rotational operator. The metaheuristic optimization based on the firefly algorithm is utilized to generate a dataset of viable element position vectors where the array is scanned to a specified angle. Analysis of the positional vectors shows approximately symmetrical patterns for element positions in each quadrant, with respect to the scanned angle. A robust model utilizing both the joint probability density function (pdf) of the number of elements in each quadrant along with a joint pdf of the first element's location and the total angle subtended per quadrant for activating elements along the circular array is derived and its performance demonstrated.

11:30

5aSP14. Efficient algorithm for active noise control of impulsive noise. Allina Mirza and Ayesha Zeb (NUST, Mechatronic Dept., Rawalpindi, Pakistan, ayeshazeb79@ee.ceme.edu.pk)

Active Noise Control (ANC) systems employing adaptive filters suffer from stability issues in the presence of impulsive noise. Due to its simplicity and less computational complexity, the Filtered-x Least Mean Square (FxLMS) algorithm is the most widely used ANC algorithm which minimizes the mean square error of noise, but it lacks robustness and stability in presence of high impulses. To overcome this limitation, new methods must be investigated. A robust adaptive algorithm (Bhagyashri algorithm) for impulsive noise suppression is already proposed in literature. In this paper, Bhagyashri algorithm is tested in ANC domain and thus Filtered-x Bhagyashri (FxBhag) algorithm is proposed. Computer simulations are executed to verify the enhanced performance of the FxBhag algorithm. The statistics of impulsive noise is modeled by Symmetric α-stable (SαS) distributions. The suggested solution exhibits better stability and faster convergence with almost same computational complexity as that of standard FxLMS algorithm and its variants.

FRIDAY MORNING, 8 DECEMBER 2017 — SALON E, 8:30 A.M. TO 11:00 A.M.

Session 5aUW

Underwater Acoustics: Underwater Measurements and Applications

Jason D. Sagers, Cochair
Environmental Science Laboratory, Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Michael J. Smith, Cochair
Center for Coastal and Ocean Mapping, University of New Hampshire, 664 Central Ave., A, Dover, NH 03820

Contributed Papers

8:30

5aUW1. Motion planning based on matched field processing with Bayesian filtering. Caitlin Bogdan, Kenric Nelson, Sean Andersson, and James G. McDaniel (Boston Univ., 110 Cummington Mall, Boston, MA 02215, cbogdan@bu.edu)

Matched Field Processing (MFP) has often been used with stationary arrays as a method for developing source location information, but potentially with large amounts of uncertainty. This has been overcome in some cases by using Bayesian Filtering to improve estimates over time of the source location by treating this quantity as a random variable. Here we show how these methods can be applied to an autonomous vehicle for motion planning. The vehicle moving through an acoustic field can provide spatially separated data that can be used in an MFP algorithm to generate an estimated source location, which can be integrated utilizing Bayesian Filtering. Motion planning methods are shown which aim to move the vehicle to the maxima of this probability distribution. This method is demonstrated using ocean acoustic simulations. [Work supported by Raytheon Advanced Studies Fellowship.]

8:45

5aUW2. Analysis of the radiated sound field of deep water multibeam echo sounders for return intensity calibration using an underwater hydrophone array. Michael J. Smith (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 664 Central Ave., A, Dover, NH 03820, msmith@ccom.unh.edu), Thomas C. Weber, Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), David Moretti (NUWC, Wakefield, RI), Anthony P. Lyons, and Val E. Schmidt (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Multibeam echo sounders (MBES) are tools used to gather geophysical information on the seafloor and watercolumn which are important for feature detection, identifying gas seeps, and characterizing the seafloor, among others. At high frequencies (>100 kHz), MBES can be calibrated for their ensonification patterns in test tanks. However, deep water MBES feature long transmit arrays and varying geometries that make tank calibration impractical. The transmit arrays can be over 8m and have a far field range in the hundreds of meters. In addition, these systems use beam steering techniques to segment the swath into multiple sectors to mitigate ship motions, which complicates the radiated pattern and return intensity. This study will better characterize the radiated sound field of deep water MBES for return intensity calibration. A MBES survey was conducted using a Kongsberg EM122 MBES on the SCORE range, a submerged broadband hydrophone array. Hydrophones were spaced ~5 km apart and were continuously recording during the survey. The EM112 is a multisector dual swath system operated at 12 kHz with CW waves. Hydrophone data were analyzed, and the resultant radiated sound field was determined at different distances and angles.

9:00

5aUW3. Spatial resolution of time-reversal processing with a virtual source array. Donghyeon Kim, Gihoon Byun (Dept. of Convergence Study on the Ocean Sci. and Technol., KIOST-KMOU Ocean Sci. and Technol. School, 727 Taejong-ro, Yeongdo-Gu, 253, Ocean Sci. and Technol., Korea Maritime and Ocean Univ., Busan Asikrks012Busan, South Korea, donghyeonkim@kmou.ac.kr), and Jea Soo Kim (Dept. of Ocean Eng., Korea Maritime and Ocean Univ., Busan, South Korea)

Time-reversal processing (TRP) can be implemented spatially and temporally to refocus incident field back to its origin. The main limitation to TRP is that it requires a probe source. This limitation was partially relaxed by the concept of a virtual source array (VSA) [S. C. Walker *et al.*, *J. Acoust. Soc. Am.* **125**, 3828–3834 (2009)] proposed for focusing time-reversed field back to the selected location without a probe source. In the case of spatial resolution of TRP, it is mathematically well documented [S. Kim *et al.*, *J. Acoust. Soc. Am.* **110**, 820–829 (2001)]. In this study, we investigate variable factors affecting spatial resolution of TRP based on a VSA. Numerical simulation results are presented and discussed.

9:15

5aUW4. Passive underwater acoustic markers for navigation and information encoding for high frequency sound navigation and ranging (SONAR) devices. Aprameya Satish, Brendan Nichols, David Trivett, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30313, aprameya.satish@gatech.edu)

AUV navigation requires accurate positioning information from the surrounding environment. Currently, several underwater navigation and surveying paradigms employ active transponders that assist in triangulation. These systems are expensive, require maintenance, and additional power sources. This paper presents a novel system that may be implemented to guide AUVs equipped with high frequency SONAR, using passive acoustic tags that are cost effective, and simple to deploy. The acoustic tags are constructed by layering multiple sheets of different acoustically reflective materials. When a deployed tag is ensonified by an encoded signal sent from a collocated source/receiver pair, the backscattered waveform by the tag yields a time-domain signature unique to the properties of the materials and geometry of the tag. The signature can be used to locate and identify any given tag within a known library of various tag designs. Numerical simulations and experimental results will demonstrate the feasibility of the proposed passive acoustic tag approach. The developed acoustic tags may find application in AUV navigation and docking exercises, swarm AUV vessel identification, AUV route planning, tagging undersea ecosystem boundaries, etc.

9:30–9:45 Break

9:45

5aUW5. Method for underwater impulsive sound characteristic measurement in non-anechoic pool. Junming Zhang, Rui Tang, Zirong Gao, and Xinyue Yu (Harbin Eng. Univ., Nantong Str. Nangang Dist. No.145, Harbin 150001, China, tangrui@hrbeu.edu.cn)

A method for measuring the sound power of impulsive sound sources in non-anechoic pool was proposed in this paper, which can also be applied to measure the sound power of other transient sound sources. The characteristic of sound field in enclosed space were presented based on normal-wave theory. By calculating the volume integral of square pressure within local sound field, the interference caused by boundary was eliminated and the steady sound power density of the enclosed space was obtained. The sound power of the source can then be obtained by introducing the sound field transmission relationship from enclosed space to free field. The method was extended to measure the sound power of transient sound sources in non-anechoic pool. The sound power of impulse sound sources was measured by using the method proposed in this paper. The effectiveness of the proposed method was proved by comparing the sound power measured in anechoic pool and non-anechoic pool.

10:00

5aUW6. Rapid measurement method of hydrophone in reverberation pool. Jundong Sun (Harbin Eng. Univ., Harbin, Heilongjaing Nantong St., China, sunjundong@hrbeu.edu.cn)

In this paper, a new method based on spatial averaging is proposed to measure hydrophone rapidly in reverberation pool. The method is to overcome the influence of acoustic normal mode by spatial averaging technique, get the reverberation sound RMS pressure in reverberation control area. It is according to the comparison principle to calibration of hydrophone rapidly. The tank is $15\mathrm{m}\times9\mathrm{m}\times6\mathrm{m}$, in the frequency range of 400–10 kHz. The free field voltage sensitivity of the four hydrophones is measured. The measurement results are compared with the free field comparison method of measurement results, the deviation is less than 1 dB, to verify the effectiveness of the method. The method can realize the simultaneous measurement of multiple hydrophones, improve the calibration efficiency, and the lower limit of measurement frequency is much lower than the lower limit frequency of the conventional pulse comparison technique, which improves the measurement capability of the anechoic tank.

10:15

5aUW7. Methods of unwrapping phase ambiguity and selecting direct sounds in an ultra short baseline positioning system. Xuyan Liu, Nan Zou, and Yifeng Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin, Heilongjiang 150000, China, liuxuyan918@126.com)

This paper presents a novel method to unwrap phase ambiguity and an expert system to select the direct sound from several reflecting pulses in an USBL positioning system. Generally, in the crossed-shaped USBL array, unwrapping ambiguity in the long edge is based on the absence of ambiguity in the short edge, where the array spacing is less than half-wavelength. However, if the positioning system works at a high frequency, which results in the short-edge phase ambiguity, the classical method to unwrap long-edge ambiguity will fail. This paper proposes a sector searching method for this scenario. It utilizes all combinations of two edges to solve all possible target orientations, and takes the mode as the final result. Additionally, hydrophones receive multi-path reflections besides the direct sound in practical application. The expert system is particularly used for selecting the direct sound correctly, which seriously effects the positioning precision. It contains series of criteria to evaluate the quality of each pulse and considers the pulse with the best quality as the direct sound. Simulation and field

experiment results prove that the proposed methods are feasible and can achieve good accuracy. Keywords: USBL; phase ambiguity; expert system; and direct sound.

10:30

5aUW8. Underwater non-cooperative communication signal recognition with deep learning. Cheng LI (Harbin Eng. Univ., No. 145 in NanTong St., Harbin, HeiLongjiang 150001, China, lichengong1@163.com), Qiming Zhou, Xiao Han, Jingwei Yin, and Mengqi Shao (Harbin Eng. Univ., Harbin, Heilongjiang Province, China)

The channel of underwater acoustics is time-varying and space-varying, which leads to the severe interference in the underwater acoustics communication signals. That makes a big challenge to distinguish the underwater acoustics communication signals modulation types. The traditional methods of identifying the non-cooperative signal modulation rely on statistics. They regard the statistical parameters in time-domain or frequency-domain of the signals as the features. In order to avoid extracting the features artificially we utilize the deep learning method to distinguish the raw time-domain signals into different modulation types such as Binary Phase Shift Keying (BPSK), Quadrature Phase Shift Keying (QPSK), 8 Phase Shift Keying (8PSK), Direct Spread Spectrum Sequence (DSSS) and Orthogonal Frequency Division Multiplexing (OFDM). The result of recognition for the experimental data shows the validity of our method. Finally we achieve 100% accuracy rate to BPSK, QPSK, DSSS modulation types and 90% accuracy rate to 8PSK, OFDM modulation types. Comparing with the statistics method, the accuracy of our method is higher.

10:45

5aUW9. An underwater measurement and control network centralized data fusion localization algorithm based on Chan-algorithm method. Tianbai Zhao, Jin Fu, Yan Wang, and Boxuan Zhang (Underwater Acoust. Eng. College, Harbin Eng. University, Rm. 914, No. 145, Nantong St., Harbin 150001, China, 627161473@qq.com)

In order to exploit the development potential of current measurement and control equipment and to build an omnibus underwater measurement and control network with higher precision, according to the working characteristics of the system itself, an underwater measurement and control network centralized data fusion localization algorithm based on Chan-algorithm method is proposed. It is an algorithm that coarsely calculates the target position according to the time of arrival by using the weighted least squares estimation(WLS) method at first; second, constructs new error vectors on the basis of the relationship between the position and the time delay information; last, resolves the target position from the vectors above by using WLS method again. The result of research demonstrates that the algorithm realizes the data fusion method of multiple sets of underwater measurement and control equipment, which could improve the precision of localization globally. The precision of the proposed method is much better than the data fusion localization algorithm purely based on the time of arrival.

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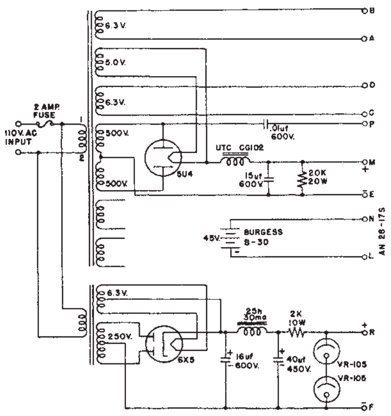

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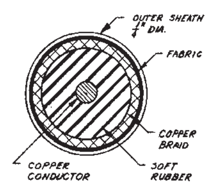

Fig.5. Schematic section showing cable makeup.

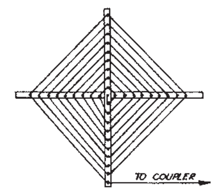

Fig.6. Configuration of the spiral mounting for the cable.

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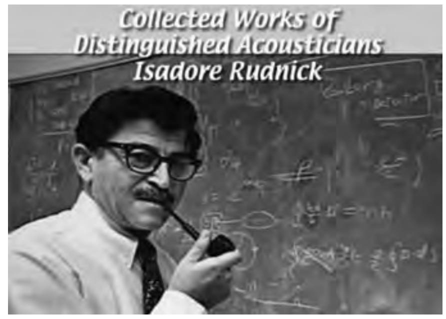
Collected Works of Distinguished Acousticians

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.

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