JASA
THE JOURNAL OF THE
ACOUSTICAL SOCIETY OF AMERICA

176th Meeting
of the Acoustical Society of America
and 2018 Acoustics Week in Canada

Vol. 144 • No. 3 • Pt. 2 of 2 • 09.2018

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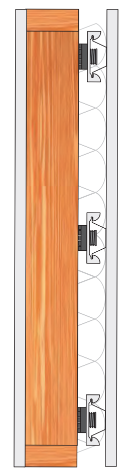

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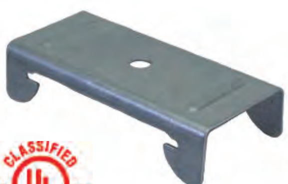

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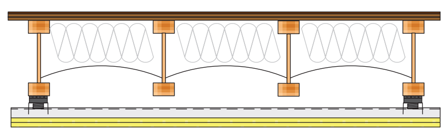

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IIC: 58

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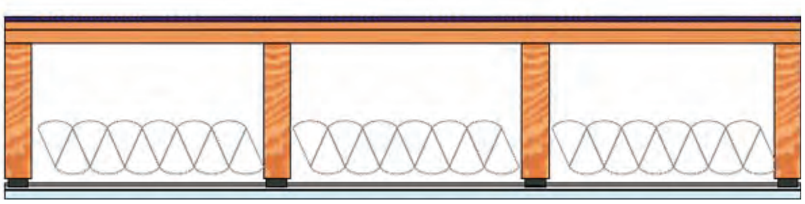

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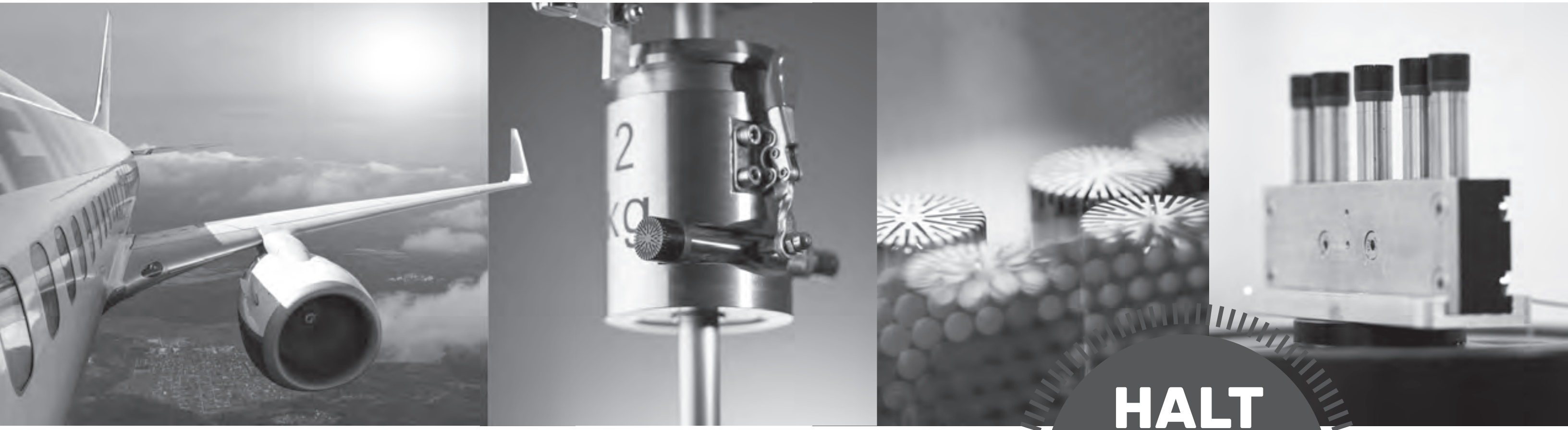


TECHNICAL PROGRAM SUMMARY

***Indicates Special Session**

MONDAY MORNING

*1aAA	Sustainable Acoustics in Social Space and WELL Buildings
*1aAB	Fish and Marine Invertebrate Bioacoustics I
*1aAO	Arctic Acoustical Oceanography I
*1aBA	State of the Art in Lung Ultrasound: Past, Present, and Future
*1aNS	Supersonic Jet Aeroacoustics I
*1aPAa	Acoustic Metamaterials and Super-Resolution Imaging
*1aPAb	Outdoor Sound Propagation I
*1aSAa	Advanced Modeling Techniques for Computational Acoustics
*1aSAb	Utilization of High-Speed Cameras to Measure Vibration
*1aSP	Machine Learning for Acoustic Applications I
*1aUW	Variability in Shallow Water Propagation and Reverberation I

MONDAY AFTERNOON

*1pAA	Auditorium Acoustics and Architectural Design: Challenges and Solutions I
*1pAB	Fish and Marine Invertebrate Bioacoustics II
*1pAO	Arctic Acoustical Oceanography II
*1pBA	Therapeutic Ultrasound Transducers
1pEAa	Transducer Characterization
1pEAb	General Topics in Engineering Acoustics
*1pNS	Supersonic Jet Aeroacoustics II
*1pPA	Outdoor Sound Propagation II
*1pPP	Understanding Limitations on Auditory Spatial Acuity
*1pSA	Advances in Thermoacoustics
*1pSCa	Coupling Phonetics and Psycholinguistics I
*1pSCb	Coupling Phonetics and Psycholinguistics II (Poster Session)
*1pSP	Machine Learning for Acoustic Applications II
1pUWa	Noise and the Environment
*1pUWb	Variability in Shallow Water Propagation and Reverberation II

MONDAY EVENING

*1eID	Tutorial Lecture on An Introduction to Sound in the Sea

TUESDAY MORNING

*2aAA	Architectural Acoustics and Audio: Even Better Than the Real Thing
2aAB	Topics in Animal Bioacoustics: Hearing
*2aAO	Machine Learning and Data Science Approaches in Ocean Acoustics I
*2aBAa	Wave Propagation in Complex Media: From Theory to Applications I
*2aBAb	Targeted Drug Delivery–Acoustic Radiation Force
*2aMU	Modeling Musical Instruments and Effects
*2aNS	Emerging Technologies for Noise Control
*2aPA	Novel Approaches to Acoustic and Elastic Wave Experimentation: Concepts, Hardware and Novel Processing Methods
*2aPP	Music, Speech, and the Brain
*2aSA	Acoustic Metamaterials
*2aSC	Recent Advances in Experimental, Computational, and Clinical Research in Voice Production and Perception
2aUW	Signals and Systems I

TUESDAY AFTERNOON

*2pAA	Auditorium Acoustics and Architectural Design: Challenges and Solutions II
*2pAB	Anything You Can Do I Can Do Better: Bat Versus Dolphin Biosonar
*2pAO	Machine Learning and Data Science Approaches in Ocean Acoustics II
2pBAa	Shock Waves and Ultrasound for Calculus Fragmentation
*2pBAb	Wave Propagation in Complex Media: From Theory to Applications II
*2pED	Measuring Educational Outcomes
*2pID	Panel Discussion: Mentoring Across Differences
*2pNS	Noise and Vibration from Fitness Activities
*2pPA	Challenges in Computational Acoustics
2pPP	Speech Perception in Children with Hearing Impairment
2pSA	General Topics in Structural Acoustics and Vibration (Poster Session)
2pSC	Speech Perception (Poster Session)
*2pUWa	Unmanned Vehicles and Acoustics
2pUWb	Signals and Systems II
2pUWc	Effects Due to Elasticity and Interfaces

WEDNESDAY MORNING

*3aAA	Advances in the Laboratory Testing of Materials
*3aAB	Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species I
*3aAO	Arctic Acoustical Oceanography III
*3aBAa	Wave Propagation in Complex Media: From Theory to Applications III
*3aBAb	Bubble Trouble in Therapeutic Ultrasound I
*3aED	Hands-On Demonstrations
*3aMU	Percussion Instruments
*3aNS	Technological Challenges in Noise Monitoring
*3aPA	Willis Coupling in Acoustic Metamaterials
3aPP	Acoustics Bricolage (Poster Session)
*3aSC	The Sound of Emotion
*3aUWa	Biological Effects on Seabed Geoacoustic Properties
3aUWb	Topics and Modelling in Underwater Acoustics

WEDNESDAY AFTERNOON

3pAA	Absorption, Diffusion, and Insulation
*3pAB	Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species II
*3pBAa	Bubble Trouble in Therapeutic Ultrasound II
*3pBAb	Wave Propagation in Complex Media: From Theory to Applications IV
*3pEA	Acoustic Particle Velocity Sensors, Algorithms, and Applications in Air
*3pEDa	Undergraduate Research Exposition (Poster Session)
3pEDb	General Topics in Acoustics Education
*3pID	Hot Topics in Acoustics
3pNS	Wind Turbine Noise
3pPP	Pitch and Sound Localization
*3pSA	History of Computational Methods in Structural Acoustics and Vibration
3pSC	Second Language Speakers and Listeners (Poster Session)
3pSPa	General Topics in Signal Processing I (Poster Session)
*3pSPb	Geometric Signal Processing in Acoustics

WEDNESDAY EVENING

*3eED	Listen Up and Get Involved

THURSDAY MORNING

*4aAA	Microphone Array Applications in Room Acoustics
4aAB	Soundscapes, Noise, and Methods in Animal Bioacoustics
4aBA	Biomedical Acoustics I
4aMU	General Topics in Musical Acoustics
*4aNS	Effects of Noise on Human Performance I
*4aPA	Interactions of Sound Beams with Objects I
*4aPP	Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations I
4aSC	Speech Production (Poster Session)
*4aSP	Detection and Tracking of Mobile Targets I
*4aUWa	Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods I
*4aUWb	Acoustic Vector Field Studies I

THURSDAY AFTERNOON

*4pAA	Validation of Modeling and Analysis: Predictions and Outcomes
*4pAB	Combining Passive and Active Acoustics for Ecological Investigations
4pAO	Topics in Acoustic Oceanography
4pBA	Biomedical Acoustics II

*4pMU	Computational and Experimental Investigations of Flow in Musical Instruments
*4pNSa	Effects of Noise on Human Performance II
4pNSb	Soundscapes
*4pPA	Interactions of Sound Beams with Objects II
*4pPP	Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations II
*4pSCa	Phonetics of Under-Documented Languages I
*4pSCb	Phonetics of Under-Documented Languages II (Poster Session)
*4pSP	Detection and Tracking of Mobile Targets II
*4pUWa	Acoustic Vector Field Studies II
*4pUWb	Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods II

FRIDAY MORNING

*5aAA	Session in Memory of Murray Hodgson I
5aAB	Marine Mammal Bioacoustics I
*5aAOa	Ocean Observatories: Laboratories for Acoustical Oceanography I
*5aAOb	Experimental Assessment of Theories of Sound Propagation in Sediments I
5aPA	General Topics in Physical Acoustics I
5aSC	Development & Clinical Populations (Poster Session)
5aSP	General Topics in Signal Processing II
*5aUW	Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods III

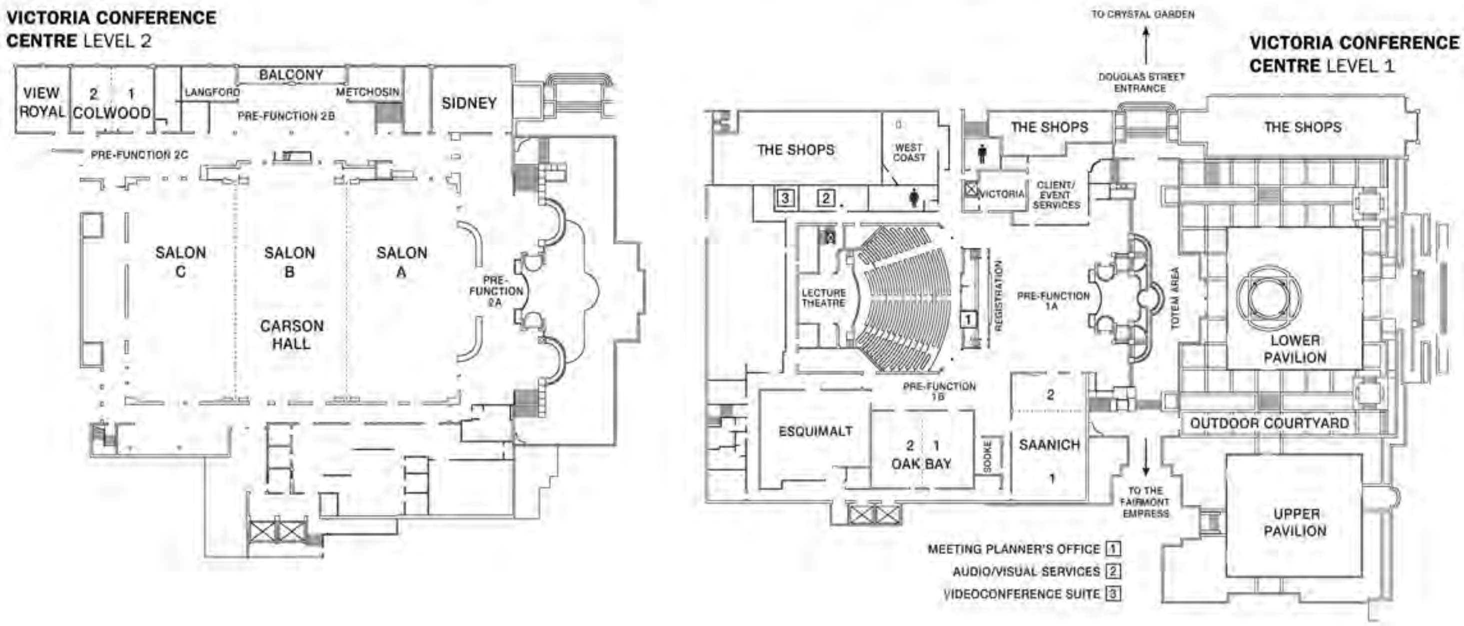
FRIDAY AFTERNOON

*5pAA	Session in Memory of Murray Hodgson II
5pAB	Marine Mammal Bioacoustics II
*5pAOa	Experimental Assessment of Theories of Sound Propagation in Sediments II
*5pAOb	Ocean Observatories: Laboratories for Acoustical Oceanography II
5pPA	General Topics in Physical Acoustics II
5pSP	General Topics in Signal Processing III

SCHEDULE OF STARTING TIMES FOR TECHNICAL SESSIONS AND TECHNICAL COMMITTEE/TECHNICAL SPECIALITY GROUP (TC/TSG) 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	Friday PM
Colwood 1/2 (VCC)		1pPP 2:00		2aBAa 8:00	2pBAb 1:00	TCPA 7:30	3aBAa 8:15	3pBAb 1:00 3pEDb 2:30	TCSP 7:30	4aPA 8:30	4pPA 1:00		5aPA 9:00	5pPA 1:00
Crystal Ballroom (FE)	1aAB 8:45	1pAB 1:00		2aMU 9:00	2pID 1:00 2pUWc 2:30		3aMU 9:45	3pID 1:00		4aMU 8:55	4pMU 1:30	TCMU 7:30		
Esquimalt (VCC)	1aAO 8:00	1pAO 1:00		2aAO 8:00	2pAO 1:00	TCAO 7:30	3aAO 8:00	3pEA 1:00			4pAO 1:00	TSGCA 4:30	5aAOb 8:00	5pAOa 1:00
Oak Bay 1/2 (VCC)	1aUW 8:00	1pUWb 1:00		2aUW 8:45	2pUWb 1:30	TCAB 7:30	3aED 8:00		3eID 6:00	4aUWa 8:30	4pUWb 1:00		5aUW 8:00	
Prefunction 1B (VCC)								3pEDa 1:00						
Rattenbury A/B (FE)	1aPAa 8:15	1pEAa 1:00 1pEAb 3:00		2aSA 9:00	2pPA 1:00	TCEA 4:30	3aPA 8:30	3pSPb 1:00		4aSP 8:00	4pSP 1:00	TCUW 7:30	5aSP 9:00	5pSP 1:00
Saanich 1/2 (VCC)	1aSAa 8:00 1aSAb 10:45	1pSA 1-5		2aPA 8:15	2pUWa 1:00	TCSA 7:30	3aUWb 9:00	3pSA 1:00		4aUWb 9:00	4pUWa 1:00		5aAOa 7:45	5pAOb 1:00
Salon A (VCC)		1pPA 3:30 1pSCa 1:00	1eID 7:00	2aSC 8:15	2pED 1:00	TCSC 7:30	3aSC 8:00				4pSCa 1:15			
Salon B (VCC)	1aPAb 9:00	1pNS 1:00		2aPP 9:00	2pPP 1:00	TCPP 7:30	3aUWa 8:30	3pPP 1:00		4aPP 9:15	4pPP 1:00			
Salon C (VCC)	1aNS 7:50	1pUWa 1:00		2aNS 8:00	2pNS 1:00		3aNS 8:00	3pNS 1:00		4aNS 8:00	4pNSa 1:00 4pNSb 2:45			
Shaughnessy (FE)	1aSP 8:00	1pSP 1:00		2aAB 9:00	2pAB 1:00		3aAB 9:00	3pAB 1:00		4aAB 8:00	4pAB 1:30	TCNS 7:30	5aAB 8:45	5pAB 1:00
Sidney (VCC)	1aBA 8:45	1pBA 1:00		2aBAb 9:15	2pBAa 1:00		3aBAb 9:00	3pBAa 1:00	TCBA 7:30	4aBA 9:15	4pBA 1:30			
Theater (VCC)	1aAA 8:00	1pAA 1:00		2aAA 8:00	2pAA 1:00	TCAA 7:30	3aAA 7:55	3pAA 1:00		4aAA 7:45	4pAA 1:15		5aAA 8:30	5pAA 1:15
Upper Pavilion (VCC)		1pSCb 3:30			2pSA 1:00 2pSC 2:00		3aPP 8:00	3pSC 1:00 3pSPa 1:00		4aSC 9:00	4pSCb 3:00		5aSC 8:30	

VICTORIA CONFERENCE CENTRE



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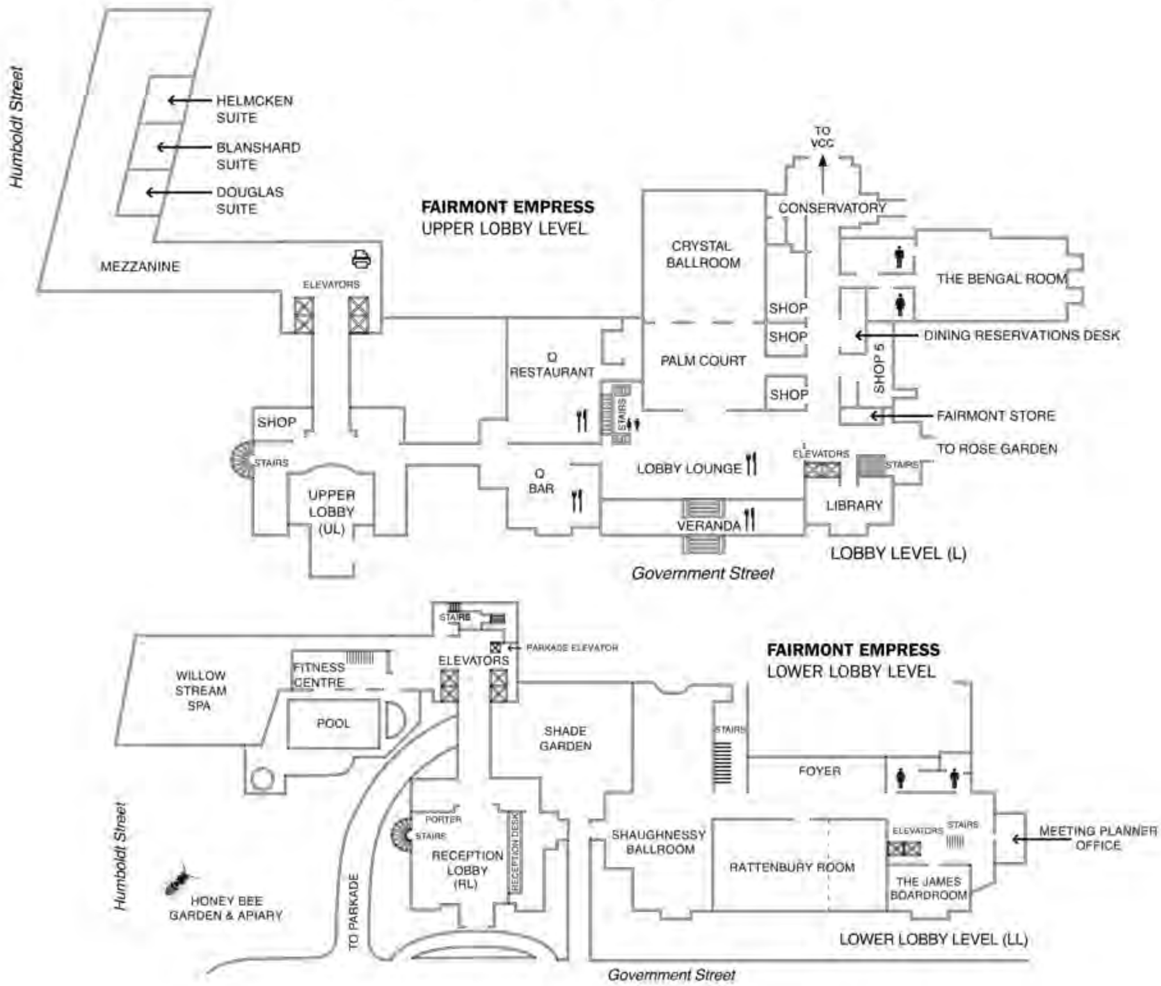

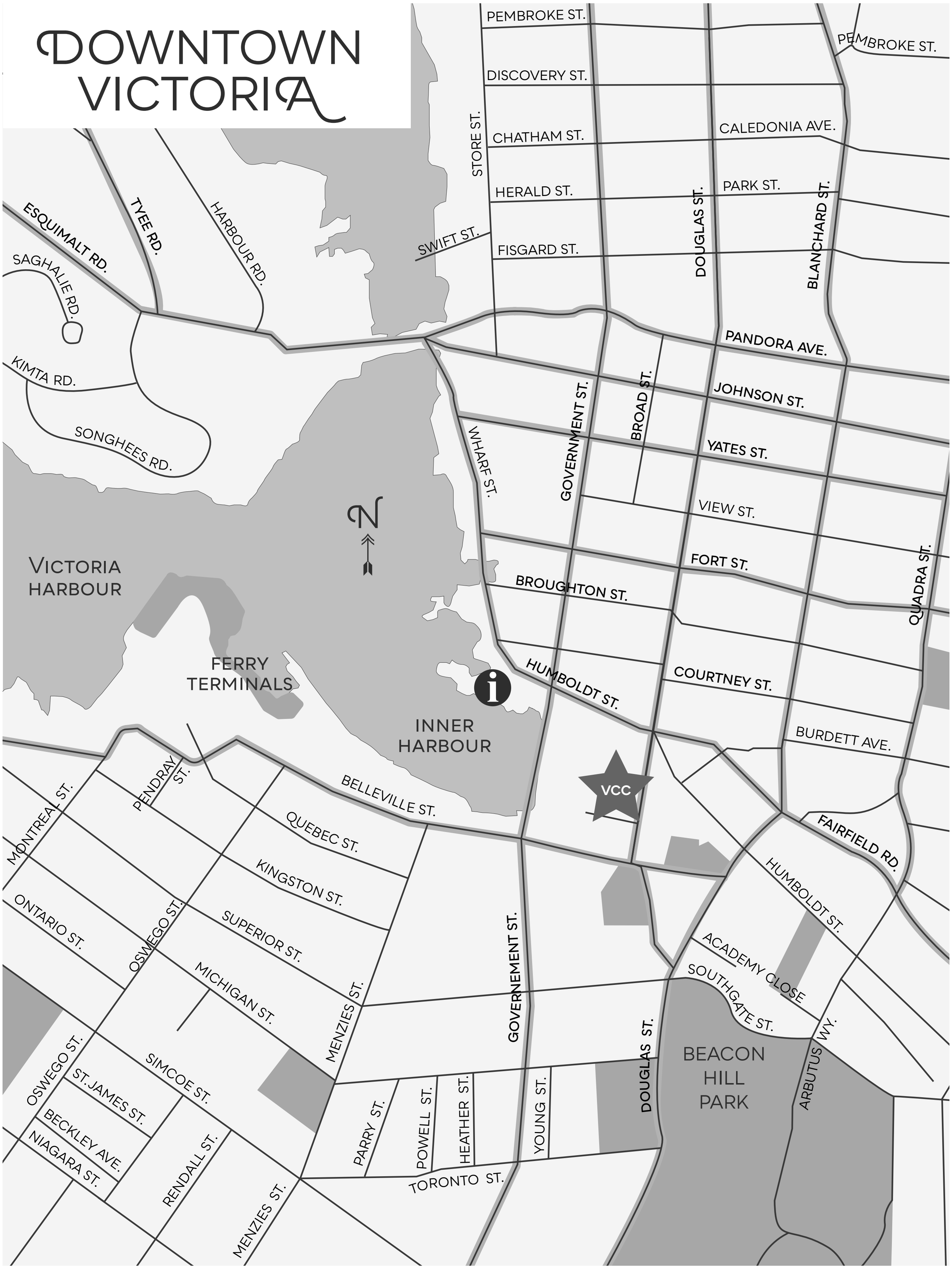
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TECHNICAL PROGRAM CALENDAR
176th Meeting of the Acoustical Society of America
and
2018 Acoustics Week Canada

MONDAY MORNING

8:00 1aAA **Architectural Acoustics:** Sustainable Acoustics in Social Space and WELL Buildings. Theater (VCC)

8:45 1aAB **Animal Bioacoustics:** Fish and Marine Invertebrate Bioacoustics I. Crystal Ballroom (FE)

8:00 1aAO **Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics:** Arctic Acoustical Oceanography I. Esquimalt (VCC)

8:45 1aBA **Biomedical Acoustics and Physical Acoustics:** State of the Art in Lung Ultrasound: Past, Present, and Future. Sidney (VCC)

7:50 1aNS **Noise, Physical Acoustics, Signal Processing in Acoustics, and ASA Committee on Standards:** Supersonic Jet Aeroacoustics I. Salon C (VCC)

8:15 1aPAa **Physical Acoustics and Signal Processing in Acoustics:** Acoustic Metamaterials and Super-Resolution Imaging. Rattenbury A/B (FE)

9:00 1aPAb **Physical Acoustics, Noise, and ASA Committee on Standards:** Outdoor Sound Propagation I. Salon B (VCC)

8:00 1aSAa **Structural Acoustics and Vibration, Physical Acoustics, Underwater Acoustics, and Architectural Acoustics:** Advanced Modeling Techniques for Computational Acoustics. Saanich 1/2 (VCC)

10:45 1aSAb **Structural Acoustics and Vibration, Engineering Acoustics, and Signal Processing in Acoustics:** Utilization of High-Speed Cameras to Measure Vibration. Saanich 1/2 (VCC)

8:00 1aSP **Signal Processing in Acoustics, Acoustical Oceanography, Architectural Acoustics, Musical Acoustics, Underwater Acoustics, and Noise:** Machine Learning for Acoustic Applications I. Shaughnessy (FE)

8:00 1aUW **Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Variability in Shallow Water Propagation and Reverberation I. Oak Bay 1/2 (VCC)

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics and ASA Committee on Standards:** Auditorium Acoustics and Architectural Design: Challenges and Solutions I. Theater (VCC)

1:00 1pAB **Animal Bioacoustics:** Fish and Marine Invertebrate Bioacoustics II. Crystal Ballroom (FE)

1:00 1pAO **Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics:** Arctic Acoustical Oceanography II. Esquimalt (VCC)

1:00 1pBA **Biomedical Acoustics and Physical Acoustics:** Therapeutic Ultrasound Transducers. Sidney (VCC)

1:00 1pEAa **Engineering Acoustics:** Transducer Characterization. Rattenbury A/B (FE)

3:00 1pEAb **Engineering Acoustics:** General Topics in Engineering Acoustics. Rattenbury A/B (FE)

1:00 1pNS **Noise, Physical Acoustics, Signal Processing in Acoustics, and ASA Committee on Standards:** Supersonic Jet Aeroacoustics II. Salon B (VCC)

3:30 1pPA **Physical Acoustics, Noise, and ASA Committee on Standards:** Outdoor Sound Propagation II. Salon A (VCC)

2:00 1pPP **Psychological and Physiological Acoustics:** Understanding Limitations on Auditory Spatial Acuity. Colwood 1/2 (VCC)

1:00 1pSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Advances in Thermoacoustics. Saanich 1/2 (VCC)

1:00 1pSCa **Speech Communication and Psychological and Physiological Acoustics:** Coupling Phonetics and Psycholinguistics I. Salon A (VCC)

3:30 1pSCb **Speech Communication and Psychological and Physiological Acoustics:** Coupling Phonetics and Psycholinguistics II (Poster Session). Upper Pavilion (VCC)

1:00 1pSP **Signal Processing in Acoustics, Acoustical Oceanography, Architectural Acoustics, Musical Acoustics, Underwater Acoustics,**

and Noise: Machine Learning for Acoustic Applications II. Shaughnessy (FE)

1:00 1pUWa **Underwater Acoustics:** Noise and the Environment. Salon C (VCC)

1:00 1pUWb **Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Variability in Shallow Water Propagation and Reverberation II. Oak Bay 1/2 (VCC)

MONDAY EVENING

7:00 1eID **Interdisciplinary:** Tutorial Lecture on An Introduction to Sound in the Sea. Salon A (VCC)

TUESDAY MORNING

8:00 2aAA **Architectural Acoustics:** Architectural Acoustics and Audio: Even Better Than the Real Thing. Theater (VCC)

9:00 2aAB **Animal Bioacoustics:** Topics in Animal Bioacoustics: Hearing. Shaughnessy (FE)

8:00 2aAO **Acoustical Oceanography, Animal Bioacoustics, Underwater Acoustics, and Signal Processing in Acoustics:** Machine Learning and Data Science Approaches in Ocean Acoustics I. Esquimalt (VCC)

8:00 2aBAa **Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications I. Colwood 1/2 (VCC)

9:15 2aBAb **Biomedical Acoustics and Physical Acoustics:** Targeted Drug Delivery–Acoustic Radiation Force. Sidney (VCC)

9:00 2aMU **Musical Acoustics and Signal Processing in Acoustics:** Modeling Musical Instruments and Effects. Crystal Ballroom (FE)

8:00 2aNS **Noise, Physical Acoustics, Structural Acoustics and Vibration, and Architectural Acoustics:** Emerging Technologies for Noise Control. Salon C (VCC)

8:15 2aPA **Physical Acoustics and Engineering Acoustics:** Novel Approaches to Acoustic and Elastic Wave Experimentation: Concepts, Hardware, and Novel Processing Methods. Saanich 1/2 (VCC)

9:00 2aPP **Psychological and Physiological Acoustics, Speech Communication, and Musical Acoustics:** Music, Speech, and the Brain. Salon B (VCC)

9:00 2aSA **Structural Acoustics and Vibration and Physical Acoustics:** Acoustic Metamaterials. Rattenbury A/B (FE)

8:15 2aSC **Speech Communication, Biomedical Acoustics, and Signal Processing in Acoustics:** Recent Advances in Experimental, Computational, and Clinical Research in Voice Production and Perception. Salon A (VCC)

8:45 2aUW **Underwater Acoustics:** Signals and Systems I. Oak Bay 1/2 (VCC)

TUESDAY AFTERNOON

1:00 2pAA **Architectural Acoustics and ASA Committee on Standards:** Auditorium Acoustics and Architectural Design: Challenges and Solutions II. Theater (VCC)

1:00 2pAB **Animal Bioacoustics and Signal Processing in Acoustics:** Anything You Can Do I Can Do Better: Bat Versus Dolphin Biosonar. Shaughnessy (FE)

1:00 2pAO **Acoustical Oceanography, Animal Bioacoustics, Underwater Acoustics, and Signal Processing in Acoustics:** Machine Learning and Data Science Approaches in Ocean Acoustics II. Esquimalt (VCC)

1:00 2pBAa **Biomedical Acoustics and Physical Acoustics:** Shock Waves and Ultrasound for Calculus Fragmentation. Sidney (VCC)

1:00 2pBAb **Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:**Wave Propagation in Complex Media: From Theory to Applications II. Colwood 1/2 (VCC)

1:00 2pED **Education in Acoustics:** Measuring Educational Outcomes. Salon A (VCC)

1:00 2pID **Interdisciplinary and Women in Acoustics:** Panel Discussion: Mentoring Across Differences. Crystal Ballroom (FE)

1:00 2pNS **Noise, Structural Acoustics and Vibration, and Architectural Acoustics:** Noise and Vibration from Fitness Activities. Salon C (VCC)

1:00 2pPA **Physical Acoustics, Architectural Acoustics, Noise, Signal Processing in Acoustics, and Underwater Acoustics:** Challenges in Computational Acoustics. Rattenbury A/B (FE)

1:00 2pPP **Psychological and Physiological Acoustics and Speech Communication:** Speech Perception in Children with Hearing Impairment. Salon B (VCC)

1:00 2pSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration (Poster Session). Upper Pavilion (VCC)

2:00 2pSC **Speech Communication:** Speech Perception (Poster Session). Upper Pavilion (VCC)

1:00 2pUWa **Underwater Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Unmanned Vehicles and Acoustics. Saanich 1/2 (VCC)

1:30 2pUWb **Underwater Acoustics:** Signals and Systems II. Oak Bay 1/2 (VCC)

2:30 2pUWc **Underwater Acoustics:** Effects Due to Elasticity and Interfaces. Crystal Ballroom (FE)

WEDNESDAY MORNING

7:55 3aAA **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration:** Advances in the Laboratory Testing of Materials. Theater (VCC)

9:00 3aAB **Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species I. Shaughnessy (FE)

8:00 3aAO **Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics:** Arctic Acoustical Oceanography III. Esquimalt (VCC)

8:15 3aBAa **Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications III. Colwood 1/2 (VCC)

9:00 3aBAb **Biomedical Acoustics and Physical Acoustics:** Bubble Trouble in Therapeutic Ultrasound I. Sidney (VCC)

8:00 3aED **Education in Acoustics:** Hands-On Demonstrations. Oak Bay 1/2 (VCC)

9:45 3aMU **Musical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Percussion Instruments. Crystal Ballroom (FE)

8:00 3aNS **Noise, ASA Committee on Standards, and Signal Processing in Acoustics:** Technological Challenges in Noise Monitoring. Salon C (VCC)

8:30 3aPA **Physical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics:** Willis Coupling in Acoustic Metamaterials. Rattenbury A/B (FE)

8:00 3aPP **Psychological and Physiological Acoustics:** Acoustics Bricolage (Poster Session). Upper Pavilion (VCC)

8:00 3aSC **Speech Communication, Musical Acoustics, and Psychological and Physiological Acoustics:** The Sound of Emotion. Salon A (VCC)

8:30 3aUWa **Underwater Acoustics, Acoustical Oceanography, Animal Bioacoustics, and Physical Acoustics:** Biological Effects on Seabed Geoacoustic Properties. Salon B (VCC)

9:00 3aUWb **Underwater Acoustics:** Topics and Modelling in Underwater Acoustics. Saanich 1/2 (VCC)

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics:** Absorption, Diffusion, and Insulation. Theater (VCC)

1:00 3pAB **Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species II. Shaughnessy (FE)

1:00 3pBAa **Biomedical Acoustics and Physical Acoustics:** Bubble Trouble in Therapeutic Ultrasound II. Sidney (VCC)

1:00 3pBAb **Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Wave Propagation in Complex Media: From Theory to Applications IV. Colwood 1/2 (VCC)

1:00 3pEA **Engineering Acoustics and Noise:** Acoustic Particle Velocity Sensors, Algorithms, and Applications in Air. Esquimalt (VCC)

1:00 3pEDa **Education in Acoustics:** Undergraduate Research Exposition (Poster Session). Prefunction 1B

2:30 3pEDb **Education in Acoustics:** General Topics In Acoustics Education. Colwood 1/2 (VCC)

1:00 3pID **Interdisciplinary:** Hot Topics in Acoustics. Crystal Ballroom (FE)

1:00 3pNS **Noise:** Wind Turbine Noise. Salon C (VCC)

1:00 3pPP **Psychological and Physiological Acoustics:** Pitch and Sound Localization. Salon B (VCC)

1:00 3pSA **Structural Acoustics and Vibration, Noise, and Signal Processing in Acoustics:** History of Computational Methods in

Structural Acoustics and Vibration. Saanich 1/2 (VCC)

1:00 3pSC **Speech Communication:** Second Language Speakers and Listeners (Poster Session). Upper Pavilion (VCC)

1:00 3pSPa **Signal Processing in Acoustics:** General Topics in Signal Processing I (Poster Session). Upper Pavilion (VCC)

1:00 3pSPb **Signal Processing in Acoustics:** Geometric Signal Processing in Acoustics. Rattenbury A/B (FE)

WEDNESDAY EVENING

6:00 3eED **Education in Acoustics and Women in Acoustics:** Listen Up and Get Involved. Oak Bay 1/2 (VCC)

THURSDAY MORNING

7:45 4aAA **Architectural Acoustics, Engineering Acoustics, Signal Processing in Acoustics, and Noise:** Microphone Array Applications in Room Acoustics. Theater (VCC)

8:00 4aAB **Animal Bioacoustics:** Soundscapes, Noise, and Methods in Animal Bioacoustics. Shaughnessy (FE)

9:15 4aBA **Biomedical Acoustics:** Biomedical Acoustics I. Sidney (VCC)

8:55 4aMU **Musical Acoustics:** General Topics in Musical Acoustics. Crystal Ballroom (FE)

8:00 4aNS **Noise, Speech Communication, and Psychological and Physiological Acoustics:** Effects of Noise on Human Performance I. Salon C (VCC)

8:30 4aPA **Physical Acoustics and Biomedical Acoustics:** Interactions of Sound Beams with Objects I. Colwood 1/2 (VCC)

9:15 4aPP **Psychological and Physiological Acoustics:** Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations I. Salon B (VCC)

9:00 4aSC **Speech Communication:** Speech Production (Poster Session). Upper Pavilion (VCC)

8:00 4aSP **Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics:** Detection and Tracking of Mobile Targets I. Rattenbury A/B (FE)

8:30 4aUWa **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics:** Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods I. Oak Bay 1/2 (VCC)

9:00 4aUWb **Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Acoustic Vector Field Studies I. Saanich 1/2 (VCC)

THURSDAY AFTERNOON

1:15 4pAA **Architectural Acoustics and Noise:** Validation of Modeling and Analysis: Predictions and Outcomes. Theater (VCC)

1:30 4pAB **Animal Bioacoustics and Signal Processing in Acoustics:** Combining Passive and Active Acoustics for Ecological Investigations. Shaughnessy (FE)

1:00 4pAO **Acoustical Oceanography:** Topics in Acoustic Oceanography. Esquimalt (VCC)

1:30 4pBA **Biomedical Acoustics:** Biomedical Acoustics II. Sidney (VCC)

1:30 4pMU **Musical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics:** Computational and Experimental Investigations of Flow in Musical Instruments. Crystal Ballroom (FE)

1:00 4pNSa **Noise, Speech Communication, and Psychological and Physiological Acoustics:** Effects of Noise on Human Performance II. Salon C (VCC)

2:45 4pNSb **Noise, and Psychological and Physiological Acoustics:** Soundscapes. Salon C (VCC)

1:00 4pPA **Physical Acoustics and Biomedical Acoustics:** Interactions of Sound Beams with Objects II. Colwood 1/2 (VCC)

1:00 4pPP **Psychological and Physiological Acoustics:** Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations II. Salon B (VCC)

1:15 4pSCa **Speech Communication:** Phonetics of Under-Documented Languages I. Salon A (VCC)

3:00 4pSCb **Speech Communication:** Phonetics of Under-Documented Languages II (Poster Session). Upper Pavilion (VCC)

1:00 4pSP **Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics:** Detection and Tracking of Mobile Targets II. Rattenbury A/B (FE)

1:00 4pUWa **Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Acoustic Vector Field Studies II. Saanich 1/2 (VCC)

1:00 4pUWb **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics:** Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods II. Oak Bay 1/2 (VCC)

8:00 5aUW **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics:** Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods III. Oak Bay 1/2 (VCC)

FRIDAY MORNING

8:30 5aAA **Architectural Acoustics and Noise:** Session in Memory of Murray Hodgson I. Theater (VCC)

8:45 5aAB **Animal Bioacoustics:** Marine Mammal Bioacoustics I. Shaughnessy (FE)

7:45 5aAOa **Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics:** Ocean Observatories: Laboratories for Acoustical Oceanography I. Saanich 1/2 (VCC)

8:00 5aAOb **Acoustical Oceanography and Underwater Acoustics:** Experimental Assessment of Theories of Sound Propagation in Sediments I. Esquimalt (VCC)

9:00 5aPA **Physical Acoustics:** General Topics in Physical Acoustics I. Colwood 1/2 (VCC)

8:30 5aSC **Speech Communication:** Development and Clinical Populations (Poster Session). Upper Pavilion (VCC)

9:00 5aSP **Signal Processing in Acoustics:** General Topics in Signal Processing II. Rattenbury A/B (FE)

FRIDAY AFTERNOON

1:15 5pAA **Architectural Acoustics and Noise:** Session in Memory of Murray Hodgson II. Theater (VCC)

1:00 5pAB **Animal Bioacoustics:** Marine Mammal Bioacoustics II. Shaughnessy (FE)

1:00 5pAOa **Acoustical Oceanography and Underwater Acoustics:** Experimental Assessment of Theories of Sound Propagation in Sediments II. Esquimalt (VCC)

1:00 5pAOb **Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics:** Ocean Observatories: Laboratories for Acoustical Oceanography II. Saanich 1/2 (VCC)

1:00 5pPA **Physical Acoustics:** General Topics in Physical Acoustics II. Colwood 1/2 (VCC)

1:00 5pSP **Signal Processing in Acoustics:** General Topics in Signal Processing III. Rattenbury A/B (FE)

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS
Fairmont Empress = (FE)/Victoria Conference Centre = (VCC)

ASA/CAA COUNCIL, BOARD OF DIRECTORS, AND ADMINISTRATIVE COMMITTEES

Sun, 4 November, 3:00 p.m.	CAA Board of Directors	Library (FE)
Sun, 4 November, 6:00 p.m.	Executive Council Dinner	Rattenbury A (FE)
Mon, 5 November, 7:30 a.m.	Executive Council	Library (FE)
Mon, 5 November, 3:30 p.m.	Technical Council	Library (FE)
Tue, 6 November, 7:00 a.m.	ASA Books	Library (FE)
Tue, 6 November, 7:30 a.m.	Panel on Public Policy	Metchosin (VCC)
Tue, 6 November, 9:45 a.m.	Spanish Speaking Acousticians	James (FE)
Tue, 6 November, 11:45 a.m.	Editorial Board	Bengal (FE)
Tue, 6 November, 12:00 noon	Student Council	Library (FE)
Tue, 6 November, 12:30 p.m.	Prizes & Special Fellowships	Douglas (FE)
Tue, 6 November, 1:30 p.m.	Meetings	View Royal (VCC)
Tue, 6 November, 4:00 p.m.	International Liaison	Douglas (FE)
Tue, 6 November, 4:00 p.m.	Newman Fund Advisory	Library (FE)
Tue, 6 November, 4:30 p.m.	Education in Acoustics	Esquimalt (VCC)
Tue, 6 November, 5:00 p.m.	Women in Acoustics	View Royal (VCC)
Wed, 7 November, 7:00 a.m.	International Research & Education	Douglas (FE)
Wed, 7 November, 7:00 a.m.	Archives & History	Metchosin (VCC)
Wed, 7 November, 7:00 a.m.	College of Fellows	James (FE)
Wed, 7 November, 7:00 a.m.	Publication Policy	Library (FE)
Wed, 7 November, 7:00 a.m.	Regional and Student Chapters	View Royal (VCC)
Wed, 7 November, 7:30 a.m.	Finance	Blanchard (FE)
Wed, 7 November, 9:30 a.m.	AS Foundation Board	James (FE)
Wed, 7 November, 11:00 a.m.	Medals and Awards	Library (FE)
Wed, 7 November, 11:30 a.m.	Public Relations	James (FE)
Wed, 7 November, 12:00 noon	Membership	Metchosin (VCC)
Thu, 8 November, 5:00 p.m.	TCAA Speech Privacy	Library (FE)
Thu, 8 November, 7:00 a.m.	Investments	James (FE)
Thu, 8 November, 7:30 a.m.	Tutorials, Short Courses, Hot Topics	Douglas (FE)
Thu, 8 November, 9:30 a.m.	Financial Affairs AC	James (FE)
Thu, 8 November, 2:00 p.m.	Strategic Plan Champions	View Royal (VCC)
Thu, 8 November, 4:30 p.m.	Member Engagement and Diversity	Library (FE)
Thu, 8 November, 4:30 p.m.	Outreach	James (FE)
Thu, 8 November, 4:30 p.m.	Publications and Standards	Blanchard (FE)
Fri, 9 November, 7:00 a.m.	Technical Council	Sidney (VCC)
Fri, 9 November, 11:00 a.m.	Executive Council	Sidney (VCC)

TECHNICAL COMMITTEE/SPECIALTY GROUP OPEN MEETINGS

Tue, 6 November, 4:30 p.m.	Engineering Acoustics	Rattenbury A/B (FE)
Tue, 6 November, 7:30 p.m.	Acoustical Oceanography	Esquimalt (VCC)
Tue, 6 November, 7:30 p.m.	Animal Bioacoustics	Oak Bay 1/2 (VCC)
Tue, 6 November, 7:30 p.m.	Architectural Acoustics	Theater (VCC)
Tue, 6 November, 7:30 p.m.	Physical Acoustics	Colwood 1/2 (VCC)
Tue, 6 November, 7:30 p.m.	Psychological and Physiological Acoustics	Salon B (VCC)
Tue, 6 November, 7:30 p.m.	Speech Communication	Salon A (VCC)
Tue, 6 November, 7:30 p.m.	Structural Acoustics and Vibration	Saanich 1/2 (VCC)
Wed, 7 November, 7:30 p.m.	Biomedical Acoustics	Sidney (VCC)
Wed, 7 November, 7:30 p.m.	Signal Processing in Acoustics	Colwood 1/2 (FE)
Thu, 8 November, 4:30 p.m.	Computational Acoustics	Esquimalt (VCC)
Thu, 8 November, 7:30 p.m.	Musical Acoustics	Crystal Ballroom (FE)
Thu, 8 November, 7:30 p.m.	Noise	Shaughnessy (FE)
Thu, 8 November, 7:30 p.m.	Underwater Acoustics	Rattenbury A/B (FE)

ASA AND CAA STANDARDS COMMITTEES AND WORKING GROUPS

Sun, 4 November, 6:30 p.m.	CAA Standards	James (FE)
Mon, 5 November, 7:00 p.m.	ASACOS Steering	James (FE)
Tue, 6 November, 7:30 a.m.	ASACOS	James (FE)
Tue, 6 November, 9:30 a.m.	S12/WG18-Room Noise	Library (FE)
Tue, 6 November, 4:30 p.m.	S3/SC1/WG 6	Blanchard (FE)
Wed, 7 November, 8:30 a.m.	S12/WG38-Product Noise Labeling	Douglas (FE)

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Sun, 4 November 1:00 p.m. - 5:00 p.m.	Short Course	Rattenbury B (FE)
Mon, 5 November 8:00 a.m. - 12:30 p.m.		Colwood 1/2 (VCC)
Mon-Thu, 5-8 November 7:30 a.m. - 5:00 p.m.	Registration	Prefunction 1A (VCC)
Fri, 9 November 7:30 a.m. - 2:00 p.m.		
Mon-Fri, 5-9 November 7:00 a.m. - 5:00 p.m.	E-mail	Prefunction 2nd fl (VCC)
Mon-Fri, 5-9 November, 7:00 a.m. - 5:00 p.m.	Internet Zone	Prefunction, 2nd fl (VCC)
Mon-Fri, 5-9 November, 7:00 a.m. - 5:00 p.m.	A/V Preview	West Coast (VCC)
Mon-Fri, 5-9 November 8:00 a.m. to 10:00 a.m.	Accompanying Persons	Palm Court (FE)
Mon-Fri, 5-9 November, 9:45 a.m. - 11:00 a.m.	Coffee Break	Prefunction 2nd fl (VCC)
Mon, 5 November, 5:00 p.m. - 5:30 p.m.	Student Orientation	Rattenbury A/B (FE)
Mon, 5 November, 5:30 p.m. - 7:00 p.m.	Student Meet and Greet	Palm Court (FE)
Mon, 5 November 5:30 p.m. - 7:00 p.m.	Early Career Speed Networking	View Royal (VCC)
Tue, 6 November, 6:00 p.m. - 7:30 p.m.	Social Hour	Royal BC Museum (across from VCC)
Wed, 7 November, 11:45 a.m. - 1:45 p.m.	Women in Acoustics Luncheon	Bengal (FE)
Wed, 7 November, 3:45 p.m. - 4:45 p.m.	Plenary Session/Awards Ceremony	Salon A (VCC)
Wed, 7 November, 4:45 p.m. - 5:30 p.m.	CAA Annual General Meeting	Salon A
Wed, 7 November, 6:00 p.m. - 8:00 p.m.	Student Reception	Shaughnessy (FE)
Wed, 7 November, 8:00 p.m. - 12:00 midnight	ASA Jam	The Sticky Wicket (see meeting information)
Thu, 8 November, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Crystal Garden (VCC)
Thu, 8 November, 6:00 p.m. - 7:30 p.m.	Social Hour	Crystal Ballroom (FE)

176th Meeting of the Acoustical Society of America
and
2018 Acoustics Week in Canada

The 176th meeting of the Acoustical Society of America/2018 Acoustics Week in Canada will be held Monday through Friday, 5-9 November 2018 at the Victoria Conference Centre and the Fairmont Empress Hotel, Victoria, British Columbia, Canada

SECTION HEADINGS

1. VICTORIA CONFERENCE CENTRE

The joint 176th Meeting of the Acoustical Society of America and 2018 Acoustics Week in Canada will be held 5-9 November 2018 at the Victoria Conference Centre and the Fairmont Empress Hotel, Victoria, British Columbia, Canada. Technical sessions, committee meetings, and other events will be held at both venues. The Social Hour on Tuesday, 6 November, will be held at the Royal BC Museum, located across from the Fairmont Empress Hotel.

2. HOTEL INFORMATION

The Fairmont Empress Hotel is the headquarters hotel where some meeting events will be held. Blocks of guest rooms at discounted rates have been reserved for meeting participants at the Fairmont Empress and the Victoria Marriott Inner Harbour Hotel.

The cut-off date for reserving rooms at special rates has passed.

Please contact the Fairmont Empress (US or Canada: 800-257-7544/All other countries: 800-441-1414 888-236-2427) a or the Victoria Marriott Inner Harbour Hotel (888-236-2427) for information about room availability.

3. TRAVELING TO CANADA

Visit the Government of Canada webstie for information about the documents you will need to enter Canada (https://www.canada.ca/en.html).

4. TRANSPORTATION AND TRAVEL

Victoria International Airport (Airport Code YYJ) is served by 14 international and domestic airlines. See http://www.victoriaairport.com for more information. The airport is approximately 15 miles (25 km) from the Victoria Conference Centre and the conference hotels in downtown Victoria.

Ground Transportation

Taxi: Taxi cabs are located curbside immediately outside of the arrivals area of the airport. Taxi rides from the airport to downtown Victoria cost about CAD $60 and take about 30 minutes. Taxis are required to accept credit card payments.

Airport Shuttle: Shuttle service is available from the airport to hotels in downtown Victoria for CAD $25 per person one-way or CAD $44 per person round-trip (2 tickets), departing every 40 minutes until 12:30 a.m. Passengers can purchase tickets at the YYJ Airport Shuttle counter in the arrivals area of the airport. See https://yyjairportshuttle.com for more information.

Car Rental: Four car rental agencies serve the airport, with counters located in the arrivals area directly across from the Information desk. See www.victoriaairport.com/car-rentals for more information.

Driving: Victoria, on Vancouver Island, can be accessed by vehicle via BC Ferries from Tsawwassen (south of Vancouver) to Swartz Bay (north of Victoria).

Parking: The cost at the Fairmont Empress Hotel is CAD $32 per day; the cost at the Marriott Inner Harbour Hotel is CAD $16 per day.

5. MESSAGES FOR ATTENDEES

A message board will be located in the Prefunction area 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.

6. 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, 5 November, at 7:30 a.m. in Prefunction 1A (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 $650 for members of the Acoustical Society of America or the Canadian Acoustical Association; $800 for non-members, $200 for Emeritus members (Emeritus status pre-approved by ASA), $375 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 and CAA Student members, $250 for students who are not members of ASA or CAA, $25 for Undergraduate Students, and $200 for accompanying persons.

One-day registration is available at $375 for members and $450 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 $800 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 2019 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 without charge. The registration fee for nonmember invited speakers who wish to participate for more than one day is $450 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.

7. ACCESSIBLITY

If you have special accessibility requirements, please indicate this below by informing 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.

8. ITINERARY PLANNER, MOBILE APP, WIFI

An Itinerary Planner and a Mobile App are available for this meeting. Please see the information on the Victoria meeting webpage for further information.

9. TECHNICAL SESSIONS

The technical program includes 121 sessions with 1282 abstracts scheduled for presentation during the meeting.

A floor plan of the Victoria Conference Centre and the Fairmont Empress 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.

10. TECHNICAL SESSION DESIGNATIONS

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

1-Monday, 5 November
2-Tuesday, 6 November
3-Wednesday, 7 November
4-Thursday, 8 November
5-Friday, 9 November

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

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

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case "a," "b," or "c" 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.

11. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 7 November, at 1:00 p.m. in the Crystal Ballroom (FE). Papers will be presented on current topics in the fields of Speech Communication, Education in Acoustics, and Animal Bioacoustics

12. TUTORIAL LECTURE ON AN INTRODUCTION TO SOUND IN THE SEA

A tutorial on “Introduction to Sound in the Sea” will be presented by Tom Dakin of Ocean Networks Canada, on Monday, 5 November at 7:00 p.m. in Salon A (VCC). 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).

13. SHORT COURSE ON BIOACOUSTICS AND ECHOACOUSTICS

A short course on Open Science and Related Topics in Hearing Research will be given in two parts: Sunday, 4 November, from 1:00 p.m. to 5:00 p.m. in Rattenbury B (FE) and Monday, 5 November, from 8:30 a.m. to 12:30 p.m. in Colwood 1/2 (VCC).

The objective of the course is to provide scholars with a solid foundation to understand bioacoustics and ecoacoustics, the equipment needed to do acoustic research and monitoring, the software tools, the applications in the different fields, ranging from basic research to environmental monitoring and protection. The course will include topics related to both terrestrial and marine bioacoustics and ecoacoustics, soundscape analysis, noise pollution, digital sound recording and analysis, also considering the importance of the acoustic environment for the human beings.

Onsite registration at the meeting will be on a space-available basis.

14. UNDERGRADUATE RESEARCH EXPOSITION

The 2018 Undergraduate Research Exposition, a poster session sponsored by Education in Acoustics will be held on Wednesday, 7 November, in session 3pEDa, in Prefunction 1B (VCC). This is a forum for undergraduate students to present their research in any area of acoustics and can also include overview papers on undergraduate research programs. It is intended to inspire and foster growth of undergraduate research throughout the Society, to encourage undergraduates to express their knowledge and interest in acoustics, and to foster their participation in the Society.

15. EARLY CAREER SPEED-NETWORKING EVENT

ASA is hosting a speed-networking event for early career participants at the meeting to facilitate professional relationships and collaboration between early career participants and more experienced members of the society. The first half of the event will include multiple short conversations between early career participants and more senior society members. The second half will be a social in which the participants will be given the opportunity to continue conversations with the more experienced society members as well as interact with other early career participants. The event will be held on Monday, 5 November, from 5:30 p.m. to 7:00 p.m. in the View Royal Room (VCC).

Participant requirements: The speed-networking event is intended for early career acousticians from any subfield of acoustics, who received their last degree within the past ten years. It is not intended for students or those in the process of receiving a degree. Students are encouraged to attend the activities specifically designed for them throughout the week. Please contact Tessa Bent (tbent@indiana.edu) or Dom Bouavichith (dbouavichith@gmail.com) if you have any questions.

16. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at the Victoria Conference Centre or the Fairmont Empress. The schedule and rooms for each Committee meeting are given on page A18.

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.

17. CANADIAN ACOUSTICAL ASSOCIATION

The Canadian Acoustical Association (CAA) is a nonprofit organization whose goals are to foster communication among people working in all areas of acoustics in Canada, promote the growth and practical application of knowledge in acoustics, and encourage education, research, protection of the environment, and employment in acoustics.

The CAA is a member society of the International Institute of Noise Control Engineering (I-INCE) and the International Commission for Acoustics (ICA), and an affiliate society of the International Institute of Acoustics and Vibration (IIAV). The Acoustical Standards Committee of the CAA also helps coordinate Canadian involvement with standards writing bodies such as ISO, IEC, ASTM, ANSI/ASA, and the Canadian Standards Association (CSA).

Each year, the CAA organizes a national conference under the banner name Acoustics Week in Canada, which is an opportunity for professional, scientific and technical exchange in all areas of acoustics and vibration. This year, it has been our pleasure to collaborate with our ASA colleagues to co-host the meeting in Victoria.

The CAA publishes *Canadian Acoustics*, a quarterly journal of refereed articles, technical notes, case studies, book reviews and news items on acoustics and vibration. A number of prizes, awards and travel grants are also offered annually to high-school, undergraduate and graduate students, postdoctoral fellows and other researchers.

More information about the CAA may be found on the association website at caa-aca.ca

18. ASA PLENARY SESSION AND AWARDS CEREMONY / CAA ANNUAL GENERAL MEETING

The ASA Plenary Session and Awards Ceremony will be held on Wednesday, 7 November, at 3:45 p.m. in Salon A at the Victoria Conference Centre.

This event will begin with a traditional First Nations welcome.

Certificates will be presented to ASA Fellows elected at the Minneapolis meeting. See page 1875 for a list of fellows.

ASA will present the Wallace Clement Sabine Medal to Michael Vorländer.

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

The CAA Annual General Meeting will be held following the plenary session at approximately 4:45 p.m. see page 1875 for the CAA Awards to be presented.

19. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be not held at the 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 A18 or on the standards bulletin board in the registration area, e.g., S12/WGI8-Room Criteria.

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: 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

20. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:45 a.m. to 10:45 a.m. in the second floor prefunction area at the Victoria Conference Centre.

21. A/V PREVIEW ROOM

The West Coast Room at the Victoria Conference Centre will be set up as an A/V preview room for authors' convenience, and will be available on Monday through Friday from 7:00 a.m. to 5:00 p.m.

22. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

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

Authors who are members of the Canadian Acoustical Association are invited to submit 2-page conference proceedings papers that will be published in Canadian Acoustics in December 2018. CAA authors who also submit a paper to POMA should change the POMA paper title to differentiate from the CAA proceedings paper. See the CAA webpage at https://awc.caa-aca.ca/index.php/AWC/awc18 for details.

23. E-MAIL AND INTERNET ZONE

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

Computers providing e-mail access will be available from 7:00 a.m. to 5:00 p.m., Monday to Friday in Prefunction area on the second floor at the Victoria Conference Centre.

Tables with power cords will also be available for attendees to gather and to power-up their electronic devices.

24. SOCIALS

Two socials with complimentary buffets and cash bars will be held on Tuesday and Thursday, 6 and 8 November, 6:00 p.m. to 7:30 p.m..

The social on Tuesday, 6 November, will be held at the Royal BC Museum across the street from the Empress Hotel. Regularly rated the best museum in Canada, the Royal BC Museum explores the natural and human history of British Columbia including natural history dioramas; recreations of a frontier town, gold mine, saw mill and fishing village; and the stunning First Peoples gallery and Totem Hall.

Please take the opportunity to circulate on the second and third floors of the museum to visit the exhibits (food and drink stations will be distributed throughout the museum).

The social on Thursday, 8 November, will be held in the Crystal Ballroom at the Empress Hotel.

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 the Technical Committees that begin immediately after the socials.

25. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 8 November, at 12:00 noon in the Crystal Garden at the Victoria Conference Centre. The speaker will be Bryan Gick (PhD, Yale U. '99), Professor and Guggenheim Fellow in the Department of Linguistics at the University of British Columbia, and a Senior Scientist at Haskins Laboratories.

His presentation is titled "If Bodies Could Talk" and will show how a deeper understanding of biological movement can help us to make sense of how sounds are produced for communication.

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 7 November. The cost is USD $30.00 per ticket.

26. STUDENT EVENTS: STUDENT ORIENTATION, MEET AND GREET, STUDENT RECEPTION

Follow the student twitter throughout the meeting @ASAStudents.

A Student Orientation will be held on Monday, 5 November, from 5:00 p.m. to 5:30 p.m. in the Rattenbury Room (FE). This will be followed by the Student Meet and Greet from 5:30 p.m. to 7:00 p.m. in the Palm Court (FE) where refreshments and a cash bar will be available.

The Students' Reception will be held on Wednesday, 7 November, from 6:00 p.m. to 8:00 p.m. in the Shaughnessy Ballroom (FE). 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.

27. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 7 November, in the Bengal Room (FE). Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 6 November. The fee is USD $30 for non-students and USD$ 15 for students.

28. JAM SESSION

You are invited to The Sticky Wicket Pub (919 Douglas St.) on Wednesday night, 7 November, from 8:00 p.m. to midnight for the ASA Jam. The Sticky Wicket Pub is a 3-5 minute walk from the Victoria Conference Centre (walk north (left) on Douglas St. for 2 blocks. The Sticky Wicket will be on the right at the corner of Courtney St. Entrance is on Courtney St.)

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.

29. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Victoria meeting. The on-site registration fee for accompanying persons is USD $200. A hospitality room for accompanying persons will be open in the Palm Court (FE) from 8:00 a.m. to 10:00 a.m. Monday through Friday. This entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

The program will include speakers on the history and culture of the city including presentations about the Emily Carr House, Butchart Gardens, Craigdarroch Castle, and St. Ann's Academy National Historic Site. Check back to the meeting website for updated information.

Victoria, the capital of British Columbia, is located on the southern tip of Vancouver Island. Centered on the bustling Inner Harbour, and graced with abundant parkland and gardens, the city is a unique blend of old world charm and new world experiences. Within a short walk of the meeting hotels is a multitude of museums and attractions as well as a broad range of culinary experiences. There are also a number of excellent tours available, including historic and cultural city sights, the world-famous Butchart Gardens, and whale-watching excursions. Tourism Victoria (https://www.tourismvictoria.com) and the Victoria Visitors Centre (812 Wharf St) are great resources.

30. WEATHER

Victoria has a Mediterranean climate with mild fall/winter weather. Average high and low temperatures in November are 50° F (10° C) and 43° F (6° C), respectively. Rainfall is common in Victoria in the fall and occurs an average of 15 days in November. Carrying a small folding umbrella may be useful for occasional showers.

31. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Roberto Racca, Chair; David P. Knobles, Acoustical Oceanography; David Mann, Animal Bioacoustics; Ian Hoffman, Shane Kanter, Architectural Acoustics; Siddhartha Sikdar, Kang Kim, Biomedical Acoustics; Benjamin Tucker, Daniel Russell, Education in Acoustics; Michael Haberman, Caleb Sieck, Engineering Acoustics; Whitney Coyle, Peter Rucz, Musical Acoustics; William J. Murphy, Kerrie Standlee, Hales Swift, Noise, Kevin Lee, Physical Acoustics; Anna Diedesch, Psychological and Physiological Acoustics; Kai Gemba, Paul Hursky, Kainam Thomas Wong, Signal Processing in Acoustics; Rajka Smiljanic, Kristin Van Engen, Speech Communication; Benjamin Shafer, Robert A. Koch, Structural Acoustics and Vibration; Jon Collis, Underwater Acoustics; Vahid Naderyan, Student Council.

32. MEETING ORGANIZING COMMITTEE

Stan Dosso, Chair; Roberto Racca, Technical Program Chair; Jorge Quijano, Graham Warner, Student Coordinators; Svein Vagle, Tom Dakin, Signs, Terry Russell, Catering; Sonya Bird, Xavier Mouy, Special Events: Shelley Dosso, Accompanying Persons;

33. PHOTOGRAPHING AND RECORDING

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

34. ABSTRACT ERRATA

This meeting program is Part 2 of the September 2018 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*.

35. GUIDELINES FOR ORAL PRESENTATIONS

- Preparation of Visual Aids See the guidelines for computer projection in section 37 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.
- Go to the A/V preview room the day before or as early as possible before your session to submit your presentation file to the AV staff.
- The A/V staff will arrange to upload your file on to the laptop in your session room.
- Arrive early enough so that you can meet the session chair, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation.
- 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.

36. 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.

37. GUIDELINES FOR USE OF COMPUTER PROJECTION

A PC computer with monaural audio playback capability and projector will be provided in each meeting room which will be used by authors who plan to use computer projection for their presentations. Authors should bring computer presentations on a CD or USB drive to load onto the provided computer and go to the A/V preview room the day before the start of their sessions. Assistance in loading presentations onto the computers will be provided in the A/V preview room.

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.

38. 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

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

Session 1aAA

Architectural Acoustics: Sustainable Acoustics in Social Space and WELL Buildings

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

Andy Chung, Cochair
HKUST, Hong Kong Plaza, Hong Kong HKSAR, Hong Kong

Ethan Bourdeau, Cochair
Standard Development, International WELL Building Institute, 381 Park Avenue South, Suite 1101, New York, NY 10016

Chair's Introduction—8:00

Invited Papers

8:05

1aAA1. Acoustical safety as important as comfort and wellness. Fumiaki Satoh (Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan, fumiaki.satoh@it-chiba.ac.jp)

Needless to say, the points of comfort and wellness are very important for the evaluation of spaces. In addition, how about the point of safety? This doesn't mean structural safety of buildings or safety to prevent crimes. For example, the luck of intelligibility of evacuation announcement in an emergency, this is an unsafe situation. It's a matter of "acoustical safety." To use absorption materials in spaces is very effective on the point of comfort and it leads to be better condition on the point of safety at the same time; however, the point of "acoustical safety" should be emphasized more. In the current situation in Japan, the consideration of using absorption materials for comfortable and safe spaces doesn't seem to be enough. This is a common understanding in our Japanese researchers on architectural acoustics. Therefore, a committee in Architectural Institute of Japan published a book in order to be popular using absorption materials. In this presentation, the contents about the book and the discussions up to publication will be introduced, in addition to some works on acoustical safety.

8:25

1aAA2. Noise analysis of urban change in the structure of health care up to the present from the beginning of the Republican era in Turkey: Case studies from Ankara Numune Hospital. Filiz B. Kocyigit, Cagri Bulhaz, and Cilga Resuloglu (İnterior Architecture, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

This research investigates the acoustical characteristics of hospitals' building environment of the Republican period and the contemporary era. Ankara Numune Hospital has been selected as a sample for research. The acoustical effect of the evolution of urban planning on the edges of Numune hospital in Capital city Ankara is analyzed. Green areas surrounded the boundaries of Numune hospital in the 1930s, whereas nowadays these green areas have been narrowed during decades. Social, political, and economic relationships changed as a result of urbanization in Turkey. Urban planning decisions affected the development, particularly the (historic) city center of Ankara where Numune hospital, the oldest hospital in Ankara, is located. Results, at the research, include equivalent sound pressure levels (Leq) as a function of location, frequency, and time of day. It appears that there are important problems in measurements taken at the entrance and at the hospital periphery and at different days and times. Average equivalent sound levels are in the 50–60 dB (A) range for 1 min, $\frac{1}{2}$, and 24 h averaging time periods. The spectra are generally flat over the 63–2000 Hz octave bands, with higher sound levels at lower frequencies, and a gradual roll off above 2000 Hz. Many units exhibit little if any reduction of sound levels in the nighttime.

8:45

1aAA3. Acoustics comfort in buildings in Hong Kong, Macao, and the Greater Bay Area of China. W. M. To (Macao Polytechnic Inst., Macao, Macao) and Andy Chung (Macau Instituto de Acustica, Macau, Macau SAR 853, China, ac@smartcitymaker.com)

There have been various assessment standards on green and sustainable buildings to promote the better use of resources following a holistic approach. As one of the major sources of complaints, noise is an essential element in these schemes. More and more smart use of materials and design embedded is seen. This paper reviews the role of sound and acoustics in the whole life cycle for a building—from planning, design to construction, and operation. A review on the latest trends and way forward will also be given.

9:05

1aAA4. Introduction to and implementation of the WELL sound concept. Ethan Bourdeau (Standard Development, Int., WELL Bldg. Inst., 381 Park Ave. South, Ste. 1101, New York, NY 10016, epbourdeau@gmail.com)

The latest revision of the WELL rating system relocates all previous acoustic related features from the Comfort concept into the new Sound concept. The new iteration of WELL reassesses the importance of all features in terms of substantiation, evidence, best practice, and feasibility, most notably for implementation within existing spaces where minimal mitigation can be anticipated. This paper introduces the new feature set of the Sound concept with a brief introduction of the new WELL rating system platform, provides an example of implementation through a case study of the new International WELL Building Institute (IWBI) headquarters, and serves as a continuation to the discussion of acoustics in ratings systems that address IEQ, health, and well-being.

9:25

1aAA5. Using the new ASHRAE 189.1 Standard for the Design of High-Performance Green Buildings—Acoustical Control Section. Erik Miller-Klein (A3 Acoust., LLP, 241 South Lander St., Ste. 200, Seattle, WA 98134, erik@a3acoustics.com) and Jeff Boldt (IMEG Corp, Madison, WI)

The 2017 ASHRAE 189.1 Standard for the Design of High-Performance Green Buildings includes a complete rewrite of the acoustical control section. This section of the standard provides both performance and prescriptive paths to acoustical compliance and permits either acoustic commissioning or visual inspections to ensure the materials and assemblies are built to optimize acoustical performance. We will outline the importance of this new standard and review the basic parameters of the acoustical performance requirements, and how this standard can be used for your projects. This session will discuss the basis for this updated section and how you can use this standard to help guide your clients, and satisfy the LEED v4 for BD + C. The 189.1/IgCC aligns well with LEED, WELL, and other green building and improves on the LEED certification standard, so basic compliance will often garner the acoustic performance points.

Contributed Paper

9:45

1aAA6. Evaluating acoustical performance of existing offices as part of a WELL feasibility service. Ryan Bessey (RWDI, 901 King St. West, Ste. 400, Toronto, ON M5V 3H5, Canada, ryan.bessey@rwdi.com) and Jessie Roy (RWDI, Calgary, AB, Canada)

While the WELL building standard is typically applied during the design of new offices, it can also be applied retroactively to existing offices. Before undertaking such an endeavour, some companies are interested in evaluating performance with respect to the standard. This paper describes how a WELL feasibility assessment can be performed for the acoustically focused WELL features. This allows companies to know where their existing space stands with respect to the WELL building standard and what it might take to achieve the different levels of certification.

10:00–10:15 Break

Invited Papers

10:15

1aAA7. Case study and lessons learned through achieving WELL Platinum. Ken Shook (Acoust., Longman Lindsey, 200 West 41st St., Ste. 1100, New York, NY 10018, kens@longmanlindsey.com)

This paper will present a case study of one of the first projects to achieve a Platinum rating through the WELL Certification program. We will review the acoustic requirement outlined by WELL and issues faced during the design of a WELL Platinum project. These will include conflicting features that were discovered between multiple features, the limitations that base-building can present, conflicts between WELL and LEED and how to satisfy both requirements. We well summarize our lessons learned and how we have applied these into recommendations to the WELL acoustic committee through our advisory role.

10:35

1aAA8. Soundscape evaluation of urban social spaces. A comparative study: Berlin-Granada. 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) and Jeronimo Vida (Dept. of Appl. Phys., Univ. of Granada, Granada, Granada, Spain)

To address the design and development of healthy built environments, noise pollution has to be taken into account, especially in Europe, where over 125 million people are affected by noise pollution from traffic every year. Since the release of the European Environmental Noise Directive in 2002, there has been a growing interest in protecting and planning quiet areas as a valid tool to reduce noise pollution. However, a common methodology to properly achieve this goal is still missing. This contribution tackles this challenge, by illustrating the implementation in Berlin and Granada of a novel participatory methodology grounded on the soundscape approach, the citizen science paradigm and a novel mobile application—the Hush City app. First, the theoretical background and methods applied are described; second, preliminary findings of the comparative assessment of "everyday quiet areas" in Berlin and Granada are illustrated. In conclusion, recommendations are provided to integrate this methodology in advanced urban planning.

10:55

1aAA9. Two case studies of acoustical design in new construction using sustainable criteria: The living building challenge and WELL building design at The Georgia Institute of Technology. Jessica S. Clements (Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

The Georgia Institute of Technology is updating their campus through a series of sustainably designed buildings to create an educational eco-commons. The ongoing renewal includes the Kendeda Building for Innovated Design and the Dalney Building. These buildings are designed to meet the International Living Future Institute's Living Building Challenge Design Standard and the International Well Building Institute's WELL certification criteria, respectively. The Kendeda building consists of classrooms, commons, offices, an auditorium, and open collaborative areas. The Dalney building is primarily office and conference spaces but includes some unique challenges for adaptable open and enclosed office areas. Substantial construction for the Kendeda building is scheduled for spring of 2019 and for the Dalney building in fall of 2019. The case studies cover the design phases of the two projects and discuss the acoustical challenges in meeting these specific designs, including credit compliance, unforeseen ramifications across disciplines, and areas of problematic interpretation.

11:10

1aAA10. Just noticeable difference of masker to enhance privacy in an open-plan office. Ainun Nadiroh and Dhany Arifianto (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, ainun12@ep.its.ac.id)

If a background conversation (masker) is easy to follow, then a worker at a workstation in an open-plan office will be distracted and annoyed. The aim of this study was if the mixture of the background conversations statistical distribution close to Gaussianity, then regardless the loudness level, the intelligibility will be decreased. On the other hand, the privacy at the workstation will be increased due to the loudness level of the background conversation. To assess the privacy level in a simulated open-plan office, we propose Just Noticeable Difference (JND) of the masker. The measurement was conducted in two workstations laboratory with 64 square meters for each workstation in which three conditions (male-female, all male, and all female speakers) of babble noise was built in one of them. The other workstation was simulated with single speech sound to observe speech privacy in the presence of the masker. We used objective measures to assess the intelligibility and the privacy of the workstation. The results suggest that the saliency of the masker depends on the fundamental frequency difference (dominant speaker). The higher saliency of the masker will cause the lower of the privacy of the other workstation.

11:25

1aAA11. Continued Summer Sound Lab Academic Research: Developing thin shell precast systems for acoustical environments. Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

This paper presentation continues progress of an active multiphase acoustical research project emphasizing implementation of adaptive interactive acoustical responses in multiuse spaces to transform unhealthy learning environments. Building upon the December 2017 paper presentation, slotted and pocketed thin shell precast systems are being explored as a viable option to decrease excessive reverberation and flutter echo while increasing clarity and speech intelligibility within a multiuse space on campus. The project, launched Spring 2017, is spanning multiple phases and disciplines as faculty and students are analyzing and validating findings in both computer and physical form. Collected data will lead researchers to design applications for adaptive and resilient learning environments with minimized acoustical distractions and improved speech intelligibility. The work presented here defines prototypical concepts to inform final full-scale built form, providing scholarly merit applicable to similar spaces and/or functions.

11:40–12:00 Panel Discussion

Session 1aAB

Animal Bioacoustics: Fish and Marine Invertebrate Bioacoustics I

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

Xavier Mouy, Cochair
JASCO Applied Sciences, 2305–4464 Markham Street, Victoria, BC V8Z7X8, Canada

Chair's Introduction—8:45

Invited Papers

8:50

1aAB1. Assessing vessel slowdown as an option for reducing acoustic masking for Arctic cod in the western Canadian Arctic. Matthew Pine (Dept. of Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, matt.k.pine@gmail.com), David E. Hannay (Jasco Appl. Sci., Victoria, BC, Canada), Stephen J. Insley (Wildlife Conservation Society Canada, Whitehorse, Br. Columbia, Canada), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), and Francis Juanes (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada)

Noise from this shipping traffic can lead to acoustic masking, reducing the ability of marine animals to detect and use biologically important sounds. Vessel-slow down may be an alternative mitigation option in regions where re-routing shipping corridors to avoid important habitat for fish and marine mammals is not possible. We investigated the potential relief in masking from a 10 knot speed reduction of container and cruise ships. Based on ambient sound measurements and real shipping data, the percentage reduction in the available listening space for fish as a container or cruise ship passes under varying speeds and ambient sound conditions was shown. The mitigation effects from slower vessels (travelling at 15 knots compared to 25 knots), in terms of auditory masking, was equal between ambient sound conditions, but not equal between the type of vessel. Slowing vessels led to a substantial decrease in the listening space reductions, with the amount of reduction varying by distance away from vessels. Vessel slowdown through sensitive habitat could be an effective mitigation strategy for reducing the extent of auditory masking.

9:10

1aAB2. Auditory threshold in fishes: Towards international standard measurement procedures. Michele B. Halvorsen (CSA Ocean Sci. Inc., 8502 SW Kansas Hwy, Stuart, FL 34997, mhalvorsen@conshelf.com) and Michael A. Ainslie (JASCO Appl. Sci., Eschborn, Germany)

Fish rely on their auditory system for survival to detect and interpret sounds in their surroundings, especially those originating from predators, prey, and conspecifics. The sensitivity of fish hearing can be impaired, for example, by masking or temporary threshold shifts. To understand and quantify the impact of noise in the oceans, we need a quantitative and comparable understanding of fishes' impaired and unimpaired hearing sensitivities to sound pressure and sound particle motion. Historically, hearing sensitivities were measured to generate an audiogram for a fish species of interest. Therefore, audiograms are only available for a few individuals representing a few species. Crucially, differences between measurement protocols leave existing audiograms mostly incomparable. The goldfish (*Carassius auratus*) is a species for which many audiograms have been measured and the reported hearing thresholds exceed a 40 dB spread, largely attributed to differences in methodology rather than in hearing sensitivity [Ladich & Fay, 2013; Maruska & Sisneros, 2016]. These large differences indicate a need to harmonize measurement procedures. Standardization of hearing sensitivity measurements is therefore essential if we are to achieve environmental management goals in the United States, Europe, and worldwide. We propose a path that leads to the development of suitable international standards.

9:30

1aAB3. Listening in the dark—Auditory evoked potential measurements and novel imaging techniques elucidate the role of hearing in deep sea fishes. Klaus Lucke (Australia, JASCO Appl. Sci., 1/14 Hook St., Capalaba, QLD 4157, Australia, klaus.lucke@jasco.com), Shaun P. Collin, Caroline Kerr, Victoria Camilieri-Asch (Oceans Graduate School and the Oceans Inst., The Univ. of Western Australia, Crawley, WA, Australia), Joanna Krieg (School of Biological Sci. and the Oceans Inst., The Univ. of Western Australia, Crawley, WA, Australia), Robert McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA, Australia), and Lucille Chapuis (Oceans Graduate School and the Oceans Inst., The Univ. of Western Australia, Crawley, WA, Australia)

In the aphotic zone of the deep ocean, mechanisms for conveying information to conspecifics or gathering information about the environment and potential predators are subject to the same physical and ecological limitations as near to the sea surface with the addition of no available light. Passive or active communication often utilises bioluminescent visual, olfactory, or acoustic signals. The sensitivity of deep sea fish to sound has never been tested. In a multi-disciplinary study, deep sea fishes were sampled in the Indian Ocean and examined for acoustic sensitivity. Acoustic signals were generated in a tubular steel tank using a calibrated sound source with pressure and particle motion of generated signals measured. The auditory sensitivity was tested using the auditory brainstem responses to amplitude modulated stimuli. To determine if the pressure or particle motion component of the acoustic signals was most relevant, specimens of all species tested were imaged in a micro-CT using a modified contrast enhancing method. This method provides increased image quality of *in-situ* morphological features, including the swim bladder and ancillary structures possibly related to for sound perception. Correlating these physiological and morphological results provides the key for an ecologically meaningful assessment of hearing capabilities of deep sea fishes.

9:45

1aAB4. Assessing ontogenetic change in acoustic pressure sensitivity in larval fishes through modelling. Andria K. Salas (Integrative Biology, Univ. of Texas at Austin, 205 W. 24th St. A6700, Austin, TX 78712, aksalas@utexas.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Lee A. Fuiman (Marine Sci. Inst., Univ. of Texas at Austin, Port Aransas, TX)

Detecting acoustic pressure can improve a fish's survival and fitness through increased sensitivity to environmental sounds. Pressure detection results from the interactions between the swim bladder and otoliths. In larval fishes, those interactions change rapidly as growth and development alter bladder dimensions and otolith-bladder distances. We used computed tomography imagery of lab-reared larval red drum (*Sciaenops ocellatus*) in a finite-element model to assess ontogenetic change in acoustic pressure sensitivity in response to a plane wave at frequencies within the range of detection by fishes. We compared the acceleration at points on the sagitta, asteriscus, and lapillus when the bladder was air-filled to results from models using a water-filled bladder. For larvae of 8.5–18 mm, the air-filled bladder amplified otolith motion by a factor of 54–3485 times that of a water-filled bladder at 100 Hz. Otolith-bladder distances increased with standard length, which decreased amplification. The concomitant rapid increase in bladder volume partially compensated for the effect of increasing otolith-bladder distances. Using idealized geometry, we found that the backbone and ribs have a negligible influence on bladder motion. Our results help clarify the auditory consequences of ontogenetic changes in swim bladder morphology and otolith-bladder relationships during larval stages.

10:00–10:15 Break

10:15

1aAB5. Experimental investigation of a robust algorithm for unambiguous source directionalization by sharks. Peter H. Rogers, James S. Martin (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, peter.rogers@gatech.edu), John C. Montgomery (Inst. of Marine Sci., Univ. of Auckland, Auckland, New Zealand), and Craig A. Radford (Inst. of Marine Sci., Univ. of Auckland, Warkworth, New Zealand)

All fish can detect the three vector components of particle motion, enabling them to determine source bearing, but particle motion alone cannot resolve which direction is towards the source and which direction is away (*i.e.,* there is a 180° ambiguity). However, fish with swim bladders can also detect acoustic pressure which enables such fish to resolve this 180° ambiguity. Proposed processing algorithms, though, often employ unrealistic restrictions such as sinusoidal time dependence and free-field conditions. We propose a robust algorithm for determining the unambiguous source direction by a fish that can detect acoustic pressure. The method is based on the time-averaged acoustic intensity vector (the time-integral of the product of pressure and particle velocity). This non-oscillatory quantity points in the direction of energy flow and hence directly away from the source. The algorithm has the advantage of being applicable to signals with any time dependence, in any propagation environment. For sharks, which do not have swim bladders, we hypothesize that the vertical component of particle acceleration can be substituted for pressure, provided that the shark is sufficiently close to the water surface. We test the validity of this hypothesis by determining the direction to the source for reef noise, a biologically relevant signal, and boat noise in a range-varying shallow water environment.

10:30

1aAB6. Assessing impacts of offshore pile driving noise on the antipredator defense and shoaling behaviors of squid (*Doryteuthis pealeii*). Ian T. Jones (Biology, Woods Hole Oceanographic Institution/Massachusetts Inst. of Technol., Woods Hole Oceanographic Inst., 266 Woods Hole Rd. MS #50, Woods Hole, MA 02543, ijones@whoi.edu), Jenni Stanley (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), and T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

Impulsive pile driving occurs during construction of marine platforms, producing intense sounds that may adversely impact animals' physiology and behavior. Little is known regarding how this noise impacts sound-sensitive invertebrates such as squid, which is surprising given squids' relative abundance and key ecological role. We quantified how a commercially important squid species (*Doryteuthis pealeii*) behaviorally responded (in a controlled environment) to pile driving sounds recorded from an offshore windfarm installation within this species' habitat. Both sound pressure and particle motion components of the sound were quantified. Fifteen-minute portions of the recordings were played to individual squid. Body pattern changes, inking, jetting, and startle responses were observed during sound exposure and all squid exhibited at least one response. These responses occurred primarily during the first few noise impulses and diminished quickly over the first minute of playback, indicating short-term habituation. Responses returned after a 24-hr rest, indicating re-sensitization. Separate experiments investigated changes in shoaling behaviors by quantifying shoal cohesion and polarity in groups of squid during ten-minute noise exposures. Rapid habituation of antipredator alarm responses and changes in shoaling dynamics may alter squids' susceptibility to predation. Noise exposure may also disrupt normal intraspecific communication and ecologically relevant behavioral responses to sounds.

10:45

1aAB7. Resilience exemplified-drum fish spawning doesn't miss a beat in the eye of a hurricane. Christopher Biggs and Brad Erisman (Marine Sci. Inst., Univ. of Texas, 750 Channelview Dr, Port Aransas, TX 78373, cbiggs@utexas.edu)

Spawning locations, timing, and periodicity are important aspects of productivity and resilience in fish because they are directly related to

reproductive success. Passive acoustic monitoring can be used to study these aspects of spawning for species that produce spawning associated vocalizations. Drumming sounds produced by spotted seatrout *Cynoscion nebulosus* during spawning have been extensively described and the sound pressure level (SPL) is correlated to the intensity of spawning. We monitored spawning in seatrout April-September 2017, which coincided with a category 4 hurricane and caused a major acute disturbance. Spawning sounds within the peak frequency bandwidth of seatrout chorusing (251-500 Hz) were observed on every day of the study with an average SPL of 121.8 (CI_{95} 121.6-122.0) dB_{rms} re: 1 μPa, which was significantly higher than the background noise level of 103.9 (CI_{95} 103.8-104.0) dB_{rms} re 1 μPa (w = 0.894, $p<0.01$). Spawning was also confirmed at two sites within the eye of the hurricane. The onset of spawning shifted 2.2 hours earlier (t = 13.91, df = 36, $p < 0.01$) for five days after the hurricane. These results illustrate that spotted seatrout are extremely resilient, which indicates a primary reason they are able to thrive in highly disturbed estuarine environments.

11:00

1aAB8. Discovery of sound in the sea: Webinars as a means of communicating current underwater acoustics research to decision makers. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Gail Scowcroft, Christopher Knowlton, Holly Morin (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI), Darlene R. Ketten (Woods Hole Oceanographic Inst., Perth, Western Australia, Australia), Arthur N. Popper (Univ. of Maryland, College Park, MD), and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Underwater sound used for anthropogenic activities is reviewed and restricted under a variety of environmental regulations. Decision makers must often synthesize rapidly new scientific research results to inform their assessments of potential impacts of proposed projects. To assist this need, the University of Rhode Island Graduate School of Oceanography has teamed with Marine Acoustics, Inc., in the Discovery of Sound in the Sea (DOSITS) project to provide accurate scientific information on underwater sound through a diversity of resources and digital platforms, including webinars. Building on the foundation of the successful 2015-2016 DOSITS webinar series and informed by the results of three international regulatory community needs assessments, the DOSITS project is hosting throughout 2018 a four-part webinar series on the fundamentals of underwater hearing and potential impacts of underwater sound on marine animals, particularly marine mammals (April–May) and fishes (November). Evaluation results from the first two webinars on marine mammals showed that 90% of survey respondents were very satisfied or satisfied with the content coverage, and 97% were extremely or very likely to attend future DOSITS webinars. The webinar approach has provided much needed on-the-job training for decision makers to effectively incorporate new scientific research into their evaluation processes.

MONDAY MORNING, 5 NOVEMBER 2018 — ESQUIMALT (VCC), 8:00 A.M. TO 12:00 NOON

Session 1aAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Arctic Acoustical Oceanography I

Peter F. Worcester, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225

Mohsen Badiey, Cochair
University of Delaware, Newark, DE 19716

Hanne Sagen, Cochair
Polar Acoustic and Oceanography Group, Nansen Environmental and Remote Sensing Center, Thormøhlensgt 47, Bergen 5006, Norway

Invited Paper

8:00

1aAO1. Toward predictive understanding of the rapidly evolving Arctic Ocean—An overview. Wieslaw Maslowski, Younjoo Lee, Jaclyn Kinney (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, maslowsk@nps.edu), Ronbert Osinski (Inst. of Oceanology Polish Acad. of Sci., Sopot, Poland), and Samy Kamal (Oceanogr., Naval Postgrad. School, Monterey, CA)

Some of the most rapid climate changes on the planet are experienced in the Arctic. In particular, the Arctic has been warming at a quicker pace than any other place on Earth, what is recognized as Arctic Amplification (AA). This warming has been most visibly manifested through a declining perennial sea ice cover, increasing the potential for its transition from the permanent toward a seasonal coverage. Those changes also affect air-sea heat fluxes and amplify ice-albedo feedback, which strongly influences ocean's absorption of solar radiation. In addition, they also alter the Arctic Ocean acoustical regime, as the thinning sea ice moves faster and deforms easier,

while its reduced coverage allows increased momentum transfer from the atmosphere to the upper ocean. This talk will provide an updated overview of the recent changes and trends in the Arctic Ocean of relevance to acoustical oceanography. We will focus on the evolution of the upper ocean stratification and water masses, mesoscale processes, and their linkages to the changing regime of the sea ice cover from multi-year to first-year sea ice. Also, the latest advancements and outstanding challenges in modeling and prediction of Arctic climate change at sub-seasonal to interannual time scales will be discussed.

Contributed Papers

9:00

1aAO2. The 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment (CANAPE): An overview. 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 (College of Earth, Atmospheric, and Ocean Sci., Oregon State Univ., Corvallis, OR), and John N. Kemp (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Arctic Ocean is undergoing dramatic changes in the ice cover and ocean structure. The 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment (CANAPE) was designed to understand the effects of changing Arctic conditions on low-frequency, long-range propagation and ambient noise. Five acoustic transceivers were deployed in a pentagon with a sixth transceiver at the center, forming an ocean acoustic tomography array with a radius of 150 km in the central Beaufort Sea. The transceivers had broadband sources centered at approximately 250 Hz located at 175-m depth in the Beaufort Duct and 15 Hydrophone Modules spanning 135 m located above the sources. A Distributed Vertical Line Array receiver with 60 Hydrophone Modules spanning 540 m was embedded within the tomographic array to provide measurements of acoustic time fronts and their fluctuations. The tomographic array was largely in open water during summer, in the marginal ice zone as it transitioned across the array during the spring and autumn, and under complete ice cover during winter. The tomographic data, together with moored data from Sea-Bird MicroCATs, Acoustic Doppler Current Profilers, ice-profiling sonars, and precision temperature sensors, will help characterize the large-scale oceanographic variability throughout the year, aiding in the interpretation the acoustic data.

9:15

1aAO3. Low–frequency acoustic transmissions under sea ice as measured in the Beaufort Sea. Matthew Dzieciuch, Peter F. Worcester (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu), John A. Colosi (Naval Postgrad. School, Monterey, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (WHOI, Woods Hole, MA), Jonathan D. Nash (Oregon State Univ., Corvallis, OR), and John N. Kemp (WHOI, Woods Hole, MA)

A tomography array was deployed in the Beaufort Sea for a year beginning in late summer 2016 during the Office of Naval Research sponsored project called the *Canada Basin Acoustic Propagation Experiment* (CANAPE.) This talk will look in detail at the propagation characteristics of broadband sound transmitted at 250 Hz over 108 km to 285 km ranges as measured on a 60 element vertical line array. Over the year, the heat content and vertical stratification of the ocean change as heat is gained and lost to the atmosphere and as the momentum transfer from the wind mixes the upper layers. The changes in heat content and stratification over the year affects the travel-time of the sound as well as its fluctuations. Examples will be shown of each. Sea-ice roughness adds scattering loss as the ice thickness increases. Thus, the seasonal ice-cover modulates the transmission loss of the acoustic paths. The losses on deep turning paths are different from the sound trapped in the surface duct. The ambient noise level also changes as the seasons progress.

9:30

1aAO4. An overview of Beaufort Sea eddies, internal waves, and spice from several recent field efforts and implications for acoustic propagation. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Murat Kucukosmanoglu (Ocean Sci., Univ. of California, Santa Cruz, Santa Cruz, CA), Peter F. Worcester, Matthew Dzieciuch (IGPP, Scripps Inst. of Oceanogr., La Jolla, CA), Andrey Y. Proshutinsky, Richard A. Krishfield (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA), and Jonathan D. Nash (Oceanogr., Oregon State Univ., Corvallis, OR)

Several recent field efforts have revealed surprisingly complex and dynamic thermohaline structure in the upper ocean of the Beaufort Sea. Solitary and compact eddies with strong temperature contrasts and currents have been observed in multiple locations and are associated with vigorous mixing, staircase structure and intrusive feature formation. While many of the eddies are primarily found in the upper 300-m of the water column, rare deep eddies with cores near 500 to 1000-m depth have also been observed. Internal waves are generally weak with energies an order of magnitude less than mid-latitude values and they show marked dominance by near inertial waves, intermittency, spatial inhomogeneity, and deviations from the Garrett-Munk model. Strong intrusive structure, termed spice, is observed in the upper 150-m of the water column and is associated with the mixed layer and eddy activity. Acoustic implications of the associated sound speed structure will be discussed.

9:45

1aAO5. Effects of range-dependent sound speed on acoustic seaglider arrivals in the Canada Basin. Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Two seagliders equipped with temperature, conductivity and pressure sensors, acoustic recorders, and acoustic Doppler current profilers were deployed in the Canada Basin as part of a large scale acoustic tomography experiment in the summer of 2016 and 2017. The seagliders received transmissions from moored acoustic sources transmitting linear frequency modulated sweeps with 100 Hz bandwidth and center frequencies on the order of 250 Hz. Sources were moored at approximately 175 m depth within an acoustic duct (sometimes referred to at the Beaufort Duct), which is present in the Canada Basin due to Pacific Winter Water, enabling acoustic transmission to long ranges. Acoustic Seagliders recorded transmissions at ranges up to 480 km. The variability and stability of the acoustic duct with range was measured by temperature and salinity profiles recorded by the gliders as they transited between moored source locations. The range dependence of this duct will be explored through oceanographic measurements made in the summer of 2016 and 2017, with particular attention paid to glider transects which roughly overlapped between the two years. The effect of the range dependent sound-speed environment on the acoustic arrivals received on the gliders will be explored through parabolic equation models and received acoustic data.

10:00

1aAO6. Overview of one year acoustical oceanography measurements on Chukchi shelf during Shallow Water Canada Basin Acoustic Propagation Experiment 2016–2017. Mohsen Badiey, Justin Eickmeier (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Altan Turgut (Naval Res. Lab, Washington, DC), Sean Pecknold (DRDC Atlantic, Dartmouth, NS, Canada), Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs, Austin, TX), Peter F. Worcester, Mathew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA), and Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada)

A multi-institutional, acoustical oceanography experiment was conducted from October 2016 through November 2017 on the Chukchi continental shelf covering 100–700 m isobaths. Parallel to a deep-water experiment conducted during the same period, the Shallow Water Canada Basin Acoustic Propagation Experiment (SW CANAPE) was designed to assess basin scale acoustic signals on the shelf region while detailed oceanographic dynamic of the shelf break region, particularly the upwelling and other dynamic of the upper 500 m water column, was measured simultaneously. Multiple arrays of oceanographic sensors including upward looking ice profiler, current profiler, temperature, conductivity, and pressure profiles measured temporal and spatial dynamics of 500 m upper ocean in connection with acoustic measurements. Distributed in a 30 km^2 area north of Barrow Alaska, vertical line arrays including an L-shaped array, covered upper 200 m of the water column. Two acoustic sources placed at 148 m and 193 m depths on the shelf emitted broadband acoustic signals in frequency bands (700–1100 Hz, and 1400–4000 Hz) along and across the shelf while the sound speed and current profile and surface ice were being measured continuously. Deep water low frequency signals were also recorded. This talk provides an overview of the SW CANAPE experiment and highlights some of the detailed measurements. [Work supported by ONR.]

10:15–10:30 Break

10:30

1aAO7. Arctic Beaufort Gyre duct transmission measurements and simulations. Timothy F. Duda, Ying-Tsong Lin, Weifeng G. Zhang (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), John A. Colosi (Naval Postgrad. School, Monterey, CA), and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Transmissions of 100–300 Hz sound over distances of 220–505 km in the Beaufort Sea area for 10 months revealed strong long-term trends and short-term variability. The sound was emitted by Canada Basin Acoustic Propagation Experiment (CANAPE) Deep Water sources, and received by CANAPE Shallow Water receivers close to the Chukchi Sea. Much of the sound that arrives at these distances is trapped in the Pacific Water duct. A dynamical model of the area driven by representative forcing, which includes eddies that have propagation consequences, is used to examine changes to the water column at small eddy scales and at broader scales that can influence the sound. Processes such as variable excitation of ducted normal modes, time-variable duct sound-speed mean profile, and coupling of ducted modes to other modes by range dependent eddy features are examined with the model. The effects of these processes are quantified and compared to field observations.

10:45

1aAO8. A year-long record of low-frequency, long-range, ducted sound propagation from the Canada 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), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA), Altan Turgut (U.S. Naval Res. Lab., Washington, DC), Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada), Ying-Tsong Lin, Andrey Y. Proshutinsky, and Richard A. Krishfield (Woods Hole Oceanographic Inst., Woods Hole, MA)

The Pacific Arctic Region, has experienced decadal changes in atmospheric conditions, seasonal sea-ice coverage, and seawater temperature. From the October 2016 to September 2017, 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 an array of hydrophones with both horizontal and vertical apertures located on the Chukchi Shelf at the 150 m isobath. The propagation distances ranged from 240 km to 520 km, and the propagation conditions changed from persistently ducted in the basin to seasonally upward refracting on the continental shelf. The water column properties and ice draft were measured by oceanographic sensors on the basin tomography moorings and by an array of oceanographic moorings on the continental shelf and slope to characterize the temporal and spatial variability of the environment. This talk examines the range-dependent measurements and explains the observed variability in the received signals through propagation modeling using the oceanographic measurements. [Work sponsored by ONR.]

11:00

1aAO9. Seabed properties at the 150 m isobath as observed during the 2016–2017 Canada Basin Acoustic Propagation Experiment. Jason D. Sagers and Megan S. Ballard (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

Seabed layering and sediment properties impact sound propagation in ocean waveguides, particularly in environments where sound propagation paths repeatedly interact with the seafloor. As part of the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE), experiments were designed to investigate seabed layering and sediment properties on the Chukchi Shelf. First, the shallow water experimental region was surveyed with a subbottom profiler to provide information about the overall sediment layering. Second, ship-radiated noise from a research vessel sailing specifically designed tracks was received on the Persistent Acoustic Observation System (PECOS). These recordings provide an opportunity for short- to mid-range geoacoustic inversion for sediment properties. Third, *in-situ* acoustic sound speed measurements were made with the Acoustic Coring System (ACS) while two to five meter long core samples were simultaneously collected. This talk presents initial findings of the seabed layering and sediment properties from these three experiments. [Work sponsored by ONR.]

11:15

1aAO10. The effects of ice cover and oceanography on medium-frequency acoustic propagation on the Chukchi Shelf. Sean Pecknold (DRDC Atlantic Res. Ctr., PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada), Ildar R. Urazghildiiev (JASCO Appl. Sci., Ithaca, New York, NY), Michelle Weirathmueller (JASCO Appl. Sci., Seattle, Washington), Mohsen Badiey (Univ. of Delaware, Newark, DE), Altan Turgut (Naval Res. Lab., Washington, DC), Jason D. Sagers, and Megan S. Ballard (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

The Canada Basin Acoustic Propagation Experiment (CANAPE) was a year-long experiment exploring the changing nature of sound propagation and ambient noise in the Arctic ocean. As part of this experiment, medium-frequency signals at 0.7–1 kHz and 1–4 kHz were transmitted by two sources on the Chukchi Shelf. One of these sources was located in an area of 150 m of water depth, approximately 350 m from a directional receiver array and 50 km from an 8-element vertical line array in a water depth of about 125 m. Oceanographic sensors were located both on the arrays and on a set of moorings on the shelf, and an ice-profiling sonar was located between the arrays about 15 km from the source. In this talk, we will focus on using the measured environmental data and propagation modeling to characterize the variability observed in the short-range and long-range received acoustic signals over the course of CANAPE.

11:30

1aAO11. Measured acoustic intensity fluctuations on the Chukchi continental shelf during Canada Basin Acoustic Propagation Experiment 2017. Mohsen Badiey, Justin Eickmeier, Lin Wan (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Altan Turgut (Naval Res. Lab, Washington, DC), Sean Pecknold (DRDC Atlantic Res. Centere, Dartmouth, NS, Canada), and Megan S. Ballard (Appl. Res. Labs., Austin, TX)

One of the main objectives of the Shallow Water (SW) CANAPE experiment was to gain a thorough understanding of a yearlong propagation of broadband signals from deep to shallow water with simultaneous oceanographic and acoustic measurements together along the and across the shelf break region. Using more than eleven acoustic arrays and seven oceanographic moorings in a 30 km^2 region on the Chukchi shelf this task is being done by assessing both deep water sound signatures and shallow water source transmissions. In this paper with present analysis of acoustic signals from both shallow and deep water sources on the Chukchi continental shelf for a specific time period between June and August 2017 where a 20 dB intensity drop from along-the-shelf source (S2) at 150 m water depth was observed for more than few weeks. This intensity drop is strongly correlated with occurrence of a large oceanographic event spanning the top 150 m water column due to Pacific Water outflow from Bering sea and retreat of Marginal Ice Zone (MIZ). During the same period, cross-the-shelf source (S1) was not transmitting signal but the reception from the deep water acoustic transmitters also show variability that could be correlated with the basin scale water column variability and the ice coverage. [Work supported by ONR.]

11:45

1aAO12. Beamforming shallow water propagation of mid-frequency acoustics on the Chukchi Shelf. Justin Eickmeier, Mohsen Badiey (Univ. of Delaware, 17 McCord Dr., Newark, DE 19713, jeickmei@udel.edu), Sean Pecknold (DRDC Atlantic, Dartmouth, NS, Canada), and Altan Turgut (Acoust. Div., Naval Res. Lab, Washington, DC)

The shallow water Canadian Basin Acoustic Propagation Experiment (CANAPE 2016–2017) was designed to study the effect of oceanographic variability on the acoustic field in the Arctic. The physics of the acoustic waveguide on the Northeastern edge of the Chukchi Shelf are influenced by dynamic boundary conditions and spatio-temporal fluctuations in temperature/salinity profiles, including the upwelling of Atlantic bottom water, sinking Bering Sea surface water and sub-mesoscale eddy formation. These fluctuations influence the acoustic waveguide characteristics of a persistent sound speed channel centered at 150 m depth. The University of Delaware (UDel) deployed seven oceanographic moorings (OM) perpendicular to the isobaths on the Chukchi Shelf (145–700 m depth). A Naval Research Lab source (S1) was deployed in-line with the OMs, generating Linear Frequency Modulated (LFM) signals in alternating 700–1100 Hz and 1400–4000 Hz bands. A Vertical Line Array (VLA) was deployed in-line and within the span of the OMs. The array aperture spanned the depth of the sound speed channel. Beamforming measurements from the VLA in combination with environmental measurements and 2D PE model output will determine the degree of influence of individual physical oceanography processes on the spatio-temporal structure of the sound channel and internal acoustic wave propagation.

MONDAY MORNING, 5 NOVEMBER 2018 — SIDNEY (VCC), 8:45 A.M. TO 12:00 NOON

Session 1aBA

Biomedical Acoustics and Physical Acoustics: State of the Art in Lung Ultrasound: Past, Present, and Future

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via Sommarive, 9, Trento 38123, Italy

Martin D. Verweij, Cochair
Acoustical Wavefield Imaging, Delft University of Technology, Lorentzweg 1, Delft 2628CJ, Netherlands

Chair's Introduction—8:45

Invited Papers

8:50

1aBA1. Sonographic interstitial syndrome: Clinical aspects. Gino Soldati (Clinical and Interventional Ultrasonography Dept., Ospedale Valle del Serchio, via dell'Ospedale, 3, Castelnuovo Garfagnana, Lucca 55032, Italy, gino.soldati@uslnordovest.toscana.it), Andrea Smargiassi, and Riccardo Inchingolo (Pulmonary Medicine Dept., Fondazione Policlinico Universitario A. Gemelli IRCCS, Rome, Italy)

Lung ultrasonography is gaining ever more consideration among physicians for detecting interstitial diseases. When the tissue is aerated, ultrasound may explore only the first peripheral layer of lung. Alveolar geometry and topology interact with ultrasound like air bubbles in a foam (wet or dry). In clinical ultrasound the normal lung appears like a specular reflector. Hyperdense pre-consolidated lung is physically similar to a wet foam and is sonographically diagnosed by artifacts generically called B-Lines (Sonographyc interstitial syndrome—SIS). Many pathologic conditions are involved. In order to give more specificity to SIS and to differentiate clinical

patterns, it is crucial to analyse histologic patterns between cardiogenic and pneumogenic pathologies. Cardiogenic involvement develops through the thickening of interlobular septa, then the alveolar flooding occurs. Thickened interlobular septa can represent a specific "deterministic" acoustic trap that could generate specific artifacts. In a later stage, fluid overflows from interstitium and small free intra-alveolar bubbles are generated without structural tissue subversion. In case of pneumogenic SIS there is a structural deformation with folded and collapsed alveoli, thickened interstitium and damaged alveoli. In conclusion, specific artifacts, namely, radiofrequency signals, could be representative of specific patterns of SIS.

9:10

1aBA2. Possible role of research platforms in lung ultrasound. Piero Tortoli, Enrico Boni (Information Eng. Dept, Università di Firenze, via Santa marta 3, Firenze 50136, Italy, piero.tortoli@unifi.it), and Alessandro Ramalli (Cardiovascular Sci., KU Leuven, Leuven, Belgium)

In a few years, ultrasound research platforms, also known as open scanners, have become a great tool for facilitating the experimental activities of ultrasound labs. Ideal platforms should be easily programmed to permit visualization of the region of interest, transmission of arbitrary sequences of arbitrary waveforms, acquisition of massive amounts of raw echo-data, and, possibly, real-time implementation of innovative processing methods. Such characteristics may be particularly relevant in lung ultrasound (LUS) applications, where quantitative methods designed around the lung properties are needed. In fact, since standard US- imaging is designed to handle impedance mismatches that are typically much lower than those found in lung tissue, LUS mostly relies on the qualitative interpretation of imaging artefacts. In particular, full control of all parameters influencing the transmit/receive modalities may allow the test of original transmit/receive strategies, while the access to raw-data may permit the off-line test of novel approaches. In this talk, the main characteristics of advanced open scanners will be reviewed, and sample applications not feasible with standard clinical scanners illustrated. Emphasis will then be given to the specific contributions that such platforms can provide to LUS. Phantom and *in-vivo* results obtained with the ULA-OP research platforms will be presented.

Contributed Papers

9:30

1aBA3. Deep learning for automated detection of B-lines in lung ultrasonography. Ruud J. G. van Sloun (Eindhoven Univ. of Technol., Eindhoven, Netherlands) and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Trento 38123, Italy, libertario.demi@unitn.it)

The application of ultrasound imaging to the diagnosis of lung diseases is gaining attention. Of particular interest are several imaging-artifacts, e.g., A and B line artifacts. A-lines are hyperechoic horizontal lines, which are substantially visualized across the entire image and parallel to pleural-line. They represent the normal pattern of the lung if pneumothorax is excluded. Differently, B-line artifacts correlate with pathology and are defined as hyperechoic vertical artifacts, which originate from a point along the pleura-line and lie perpendicular to the latter. Their presence has been linked to an increase in extravascular lung water, interstitial lung diseases, non-cardiogenic lung edema, interstitial pneumonia and lung contusion. In this work, we describe a method aimed to support the clinicians by automatically identifying the frames of an ultrasound video where B-lines are found. To this end, we employ modern deep learning strategies and train a fully convolutional neural network to perform this task on b-mode images of dedicated ultrasound phantoms (Demi *et al*., Sci. Rep. 2017). We moreover calculate neural attention maps that visualize which components in the image triggered the network, thereby offering simultaneous localization. Future work includes characterization of the detected B-lines to enable adequate phenotyping of various lung pathologies.

9:45

1aBA4. Ultrasound lung spectroscopy: Preliminary *in-vivo* results. Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive 9, Trento 38123, Italy, libertario.demi@unitn.it), Marcello Demi (Medical Image Processing, Fondazione Toscana Gabriele Monasterio, Pisa, Italy), and Gino Soldati (Emergency Medicine Unit, Valle del Serchio General Hospital, Lucca, Italy)

Nowadays, lung pathologies are mainly diagnosed by X-ray imaging, with the golden-standard being computed tomography (CT). However, CT is not available at patients' bed, not always accessible, and expensive. Additionally, CT delivers a relatively high radiation-dose, with the risk of an increased possibility to develop cancer. Recently, attention is shifting toward ultrasound imaging, which would offer a more portable, more accessible, cheaper, and safer alternative. However, despite promising clinical findings, technical development is lacking. In fact, ultrasound imaging is not optimal for lung investigation being designed for low acoustical impedance mismatches. As a consequence, clinicians base their decisions on qualitative interpretations and imaging artifacts, such as the well-known B-line artifact. In this work, following a recently published *in-vitro* study (L. Demi *et al*., "Determination of a potential quantitative measure of the state of the lung using lung ultrasound spectroscopy," Scientific Reports, 2017), we report on the first preliminary *in-vivo* results. Standard B-mode, multi-frequency imaging-sequences, and RF-data, were acquired. Consistently with the *in-vitro* study, clinical data show that B-lines can be characterized on their frequency content. These results further support the hypothesis that the analysis of ultrasound spectral features can be used to develop a quantitative method dedicated to the lung.

10:00

1aBA5. Feature detection and pneumonia diagnosis based on clinical lung ultrasound imagery using deep learning. Xinliang Zheng, Sourabh Kulhare, Courosh Mehanian (Intellectual Ventures Lab., 14360 SE Eastgate Way, Bellevue, WA 98007, lzheng@intven.com), Zhijie Chen (Shenzhen Mindray Bio-medical Electronics Co, Shenzhen, China), and Ben Wilson (Intellectual Ventures Lab., Bellevue, WA)

Pneumonia is a common disease with both high morbidity and mortality. The diagnosis of pneumonia remains a clinical challenge, especially in low resource settings where prevalence is high, diagnostic devices are limited, and doctors are scarce. Lung ultrasound has been identified as a useful and low-cost tool for pneumonia diagnosis in many studies. In the present work, we first developed a convolutional neural network (CNN)-based deep learning algorithm to automatically identify four key features linked to lung conditions: pleural line, B-line, consolidation, and pleural effusion. The algorithm was trained using ultrasound data collected from over 150 pediatric and adult patients, with features annotated by expert sonographers. A single shot detection (SSD) framework was developed to detect those features in each video frame image. We then explored the accuracy of diagnosing pneumonia based on one or more lung ultrasound features, using CT as a gold standard. Our results indicate that deep learning algorithms can successfully detect abnormal lung features in ultrasound imagery, and those features can be used to diagnose pneumonia at a high sensitivity level. Computer-assisted ultrasound interpretation has the potential to place point of care, expert-level diagnostic accuracy in the hands of low-resource health care providers for pneumonia diagnosis.

10:15–10:30 Break

Invited Papers

10:30

1aBA6. Ultrasound assessment of pulmonary edema using multiple scattering. Kaustav Mohanty (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell, Behrooz Masuodi, Mir H. Ali, Thomas Egan (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., 911 oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Ultrasound imaging in the lungs is challenging because of air-filled alveoli which induce multiple scattering. Pulmonary edema is characterized by increased extravascular lung water, which causes acute dyspnea. There is an urgent need from pulmonologists to non-invasively quantify pulmonary edema, in order to monitor response to treatment. The present methodology takes advantage of multiple scattering and calculates the scattering mean free path (L*) for the quantitative assessment of pulmonary edema. L* is a semi-local property, which could be correlated to the amount of fluid buildup in the lung. Pulmonary edema was induced in 4 Sprague-Dawley rats using ischemia reperfusion injury. Following a sternotomy, L* was calculated from the time evolution of the backscattered incoherent intensity using an L11-4v linear array connected to a Verasonics scanner. Ex-vivo data were also acquired for 2 edematous pig lungs. For measurements, the left lung was edematous whereas the right lung was treated as control. Significant differences ($p<0.001$) were found between control ($L^* = 330\pm89\mu m$) and edema ($L^* = 876\pm179\ \mu m$) for *in-vivo* rats. For *ex-vivo* pig lungs, L* was found to be $190\pm84\ \mu m$ for control and $1074\pm361\ \mu m$ for edema. This suggests the potential of L* for the assessment of pulmonary edema.

10:50

1aBA7. Lung ultrasound surface wave elastography. Xiaoming Zhang (Radiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Lung ultrasound surface wave elastography (LUSWE) was developed for noninvasively measuring superficial lung tissue stiffness. In LUSWE, a 0.1 second harmonic vibration is generated on the chest wall of a subject using a handheld vibrator. An ultrasound probe is aligned with the vibration excitation in the same intercostal space to measure the generated surface wave propagation on the lung. A human subject is examined in a sitting position. The lung is tested at the total lung volume and through three intercostal spaces on each lung. In a prospective clinical study, we measure the lung surface wave speeds at 100 Hz, 150 Hz, and 200 Hz on 91 patients with interstitial lung disease (ILD) and 30 healthy control subjects. Significant differences of wave speed between patients and controls were found in all 6 lung regions and for 3 frequencies. We also found positive correlation between LUSWE and clinical assessments including CT and pulmonary function tests. LUSWE is used to track the changes of surface wave speed for patients. LUSWE is a safe and noninvasive technique for generating and measuring surface wave propagation on the lung. LUSWE may complement the clinical standard high-resolution computed tomography for assessing ILD.

Contributed Paper

11:10

1aBA8. On the artefactual information of ultrasound lung images. Marcello Demi (Medical Imaging, Fondazione Toscana Gabriele Monasterio, Via G. Moruzzi 1, Pisa 56124, Italy, demi@ftgm.it), Gino Soldati (Emergency Medicine Unit, Valle del Serchio General Hospital, Lucca, Italy), and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

In standard B mode imaging, a probe transmits brief pulses and receives the echoes reflected by every structure of the explored medium. A set of consecutive pulses are used to reconstruct a two-dimensional image even though some of the assumptions needed to do this are not exactly satisfied. Consequently, numerous visual artefacts are present in ultrasound images and physicians are aware that significant discrepancies between the ultrasound images and the anatomy of the examined medium may exist. Nonetheless, such artefacts are usually analyzed accurately since they provide clinical information and understanding of the physical mechanisms which are at the basis of their formation is important. In the case of lung images, everything we see beyond the chest wall represents artefactual information since the aerated spaces of the lung reflect most of the energy of the ultrasound pulses. Herein, we will describe the genesis of the most important lung artefacts by means of mathematical models. We will discuss how the so called A lines, B lines, and White Lung may be generated by the reverberation effects between the probe and the pleura, by the trapped and subsequently re-radiated acoustic energy and by the multiple reflections between randomly distributed aerated spaces.

Invited Paper

11:25

1aBA9. A review of pulmonary injury by diagnostic ultrasound. Douglas Miller, Zhihong Dong, Chunyan Dou (Radiology, Univ. of Michigan, 3240A Medical Science I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), and Krishnan Raghavendran (Surgery, Univ. Michigan, Ann Arbor, MI)

Diagnostic ultrasound (DUS) induction of pulmonary capillary hemorrhage (PCH) presents a unique clinical safety issue. Pulmonary DUS has become routine in clinical and point-of-care settings worldwide for diagnosis of edema, effusion, pneumonia, etc. Our experiments in rats have shown that DUS-induced PCH increased rapidly above exposure thresholds, which depended on specific conditions. Using a linear array at 6.6 MHz, peak rarefactional pressure amplitude thresholds were 1.5 MPa for B mode, 1.1 MPa for Angio Doppler, 1.1 MPa for M mode, and 0.6 MPa for pulsed Doppler. Thresholds were nearly constant from 1.5 MHz to 12 MHz, suggesting involvement of a frequency-independent mechanism, such as radiation pressure. Physiological conditions were found to be as important as physical exposure parameters. Infusion of saline reduced the effect. Xylazine in the normal ketamine/xylazine anesthesia enhanced PCH relative to ketamine only, and the clinical sedative dexmedetomidine also lowered thresholds. Rats with positive-pressure ventilation had DUS-induced PCH inhibited by only +4 cm H_2O, or enhanced by only -4 cm H_2O end expiratory pressure acting on the blood-air barrier. With subsequent studies and better understanding of this phenomenon, DUS-induced PCH will be avoidable by use of sub-threshold output by sonographers using DUS machines with output control.

Contributed Paper

11:45

1aBA10. Observation of acoustic fountain generation by diagnostic ultrasound shear wave elastography. Brandon Patterson and Douglas Miller (Radiology, Univ. of Michigan, 1301 Catherine St., Ann Arbor, MI 48109, awesome@umich.edu)

High-intensity focused ultrasound has been shown to drive fountains and atomization at liquid-gas and tissue-gas interfaces. Though these phenomena are not well studied for diagnostic ultrasound, they have been hypothesized to play a role in diagnostic ultrasound-induced pulmonary capillary hemorrhage (PCH). We demonstrate that push pulses used in diagnostic shear wave elastography (SWE) also cause fountaining and atomization at water-air and blood-air interfaces. A focused ultrasound transducer (SSI SL15-4), was aimed upward, through water, at an air interface. An SWE pulse sequence, including four push pulses (amplitude $\leq$ 8.6 MPa, pulse duration $\cong$ 650 ms, and center frequency $\cong$ 5.0 MHz), was initiated. The interface was photographed at 20k fps and four successive fountains were observed. Fountains heights up to 12 and 11 mm were observed for water and blood respectively and ejected water droplets traveled up to 30 cm above the surface. The spacing of the four fountains was measured to be the same as the spacing of four PCH areas observed on rat lungs exposed to the same pulse sequence. Concurrent studies should reveal the relative efficacy of PCH induction from SWE and normal pulse echo and Doppler imaging.

MONDAY MORNING, 5 NOVEMBER 2018 — SALON C (VCC), 7:50 A.M. TO 12:00 NOON

Session 1aNS

Noise, Physical Acoustics, Signal Processing in Acoustics, and ASA Committee on Standards: Supersonic Jet Aeroacoustics I

Allan C. Aubert, Cochair
Propulsion & Power, NAVAIR, 2344 Centennial Loop, Round Rock, TX 78665

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

Chair's Introduction—7:50

Invited Papers

7:55

1aNS1. Jet aircraft noise—Past, present, and future issues. Richard L. McKinley (Oak Ridge Inst. for Sci. and Education, 2610 Seventh St., AFRL/711HPW/RHCB, Wright-Patterson AFB, OH 45433-7901, rich3audio@aol.com) and Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Aircraft noise has been an issue since for the US Air Force since World-War I. The early days of flight had noise issues that were specific to reciprocating engines and propellers. The advent of the jet engine in the 1940s changed the noise issues and focus dramatically. The jet noise problem was first addressed in the Air Force Research Laboratory (AFRL) by Henning von Gierke and has continued until today with the development of more powerful fifth-generation fighter jet engines. This presentation will focus on the role of AFRL in noise issues with jet aircraft from the early days until now, and how that noise impacted people and communities; the research that fostered noise reductions; noise from current aircraft; the problems and issues that have persisted with jet noise; and some thoughts for future research.

8:15

1aNS2. Full-spectrum near-field acoustical holography for fighter jet noise imaging. Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Kent L. Gee, Kevin M. Leete, Mylan R. Cook (Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

The development of acoustic imaging technologies over the previous decade has proven useful for increasing our understanding of the noise generation mechanisms inside the turbulent flows of full-scale tactical aircraft engines. In particular, advancements in near-field acoustical holography have allowed for jet noise source imaging from measurements made on tied-down aircraft over hard reflecting ground surfaces. These images have been limited in bandwidth due to the spatial resolution and extent of the holography arrays. However, improved aperture extension methods have allowed for representation of the lowest frequencies, and a new tool has been developed to produce accurate sound field images at above-Nyquist frequencies for broadband sources. These two technologies are implemented together to extend the imaging of fighter jet noise near-field toward the full spectrum required for an aircraft noise model.

8:35

1aNS3. Localization of directional noise sources from high-performance military aircraft through subarray beamforming analysis. David F. Van Komen, Tracianne B. Neilsen, Blaine M. Harker, Kent L. Gee, S. Hales Swift (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), Micah Downing, Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

High-performance military aircraft noise is created by multiple sound generation mechanisms that need to be understood to guide noise reduction efforts and for adequate sound field predictions. Phased-array methods can be used to produce frequency-dependent equivalent acoustic source models. The Hybrid (beamforming) method [Padois *et al.,* J. Sound Vib. **333** (2014)] is applied to an acoustical measurement along a 71-microphone ground-based array, spanning 32 m, placed in the vicinity of a high-performance military aircraft as the engine was operated at different powers. Application of the Hybrid method to the full-array creates an overall equivalent source model that is sufficient for predicting overall field radiation but fails to separate the different noise sources. Applying the Hybrid method to subarrays separates broadband shock-associated noise from the main radiation lobes of turbulent mixing noise. Results show that the subarray-based equivalent source distributions for the different types of noise originate from overlapping source locations. Further analysis of the subarray-based equivalent noise sources using coherence and directionality from the unwrapped phase of the cross-spectral source reconstructions identifies overlapping, frequency-dependent source regions with characteristics unique to broadband shock-associated noise and turbulent mixing noise. [Work supported by an Air Force Research Laboratory SBIR.]

Contributed Papers

8:55

1aNS4. Beamforming of crackle-related events in supersonic jet noise. Aaron Vaughn, S. Hales Swift, Kent L. Gee (Brigham Young Univ., C110 ESC, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Micah Downing, Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC), and Alan T. Wall (Air Force Res. Lab., Wright-Patterson AFB, OH)

Crackle is an annoying component of supersonic jet noise. In the far field, crackle is related to the presence of acoustic shocks that develop due to nonlinear propagation; however, the intermittent source events that drive crackle generation are not well understood. This study investigates the apparent source locations of events related to crackle, which could include high-amplitude or steepened, shock-like waveforms. The measured data were obtained through ground-array measurements near a high-performance military aircraft, which was run at different engine powers. The apparent source regions corresponding to different event triggers, such as pressure amplitude, derivative amplitude, and spectral characteristics, are compared. The crackle-related event beamforming is also compared against the results of time-averaged, frequency-domain source localization. [Funded by an AFRL SBIR.]

9:10

1aNS5. Comparing two inverse array methods for source reconstructions of noise radiated from a high-performance jet aircraft. Jacob A. Ward, Kevin M. Leete, Kent L. Gee, David F. Van Komen, Tracianne B. Neilsen (Phys., Brigham Young Univ., N243 ESC, Provo, UT 84602, jacob.ward@live.com), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), Micah Downing, and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Because direct flow measurements of tactical aircraft jet engines are not currently possible, acoustic source characteristics are instead inferred from array processing. This paper compares two array processing methods using the same array data from a high-performance military aircraft. Hybrid beamforming (HBF) and multisource statistically optimized near-field acoustical holography (M-SONAH) have both been used previously for frequency-dependent jet noise source characterization, but are compared here for the same input data. A 71-element linear array of equally spaced microphones was placed approximately parallel to the shear layer covering a distance of 32 m. Complex pressures obtained from this array served as the input to both methods. Favorable agreement in terms of maximum source location, source shape, and source extent was seen between the two methods' respective results. While the methods continue to have their relative strengths and reasons for use, this favorable agreement indicates that an improved understanding of military jet noise sources has been achieved. [Work supported by an AFRL SBIR.]

9:25

1aNS6. Meteorological effects on long-range nonlinear propagation of jet noise from a static, high-performance military aircraft. Brent O. Reichman, Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), and Alan T. Wall (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

The impact of nonlinearity on the propagation of noise from military jet aircraft has been fairly well documented, but only within a few hundred meters from the aircraft. This paper describes analysis of nonlinear propagation for morning static runups of F-35 aircraft at greater distances, out to 1220 m near the direction of maximum radiation and at heights ranging from 0 m up to 30.5 m. A comparison of overall levels with distance and height reveals evidence of significant atmospheric refraction effects, and a general trend of decreasing level with height. Examination of nonlinearity metrics reveals opposite behavior, however. At these distances, nonlinear propagation effects are often strongest in waveforms with lower sound levels, which is counterintuitive. One important finding, however, is that acoustic shock strength can vary greatly from runup to runup, even for seemingly small changes in atmospheric conditions. This analysis demonstrates the need for further research into long-range nonlinear propagation of jet noise through realistic atmospheric conditions.

9:45–10:00 Break

10:00

1aNS7. Large-eddy simulation of supersonic jet noise emanating from an F404 nozzle at model scale. Junhui Liu and Ravi Ramamurti (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, junhui.liu@nrl.navy.mil)

Large-eddy simulations of supersonic jet noise emanating from an F404 nozzle at the model scale have been carried out using the JENRE flow solver. A wall-model method that was previously validated for a high subsonic flow over a flat plate is used to model the boundary layer effect inside the nozzle. The nozzle geometry is a faceted bi-conic convergent-divergent nozzle with a design Mach number equal to 1.65 and the nozzle-exit diameter equal to 5.07 inches. Both mildly and highly overexpanded conditions are tested for heated jets. The time averaged flow field, turbulence intensities, and the far-field noise are compared with available experimental data. The effects of both the boundary-layer thickness and turbulence intensity at the nozzle exit are investigated to assess their impact on jet noise generation and the noise source characteristics.

10:20

1aNS8. Sensitivity of acoustic predictions on the mesh size of large-eddy simulation of supersonic jet. Inman Jang, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, ohm@yonsei.ac.kr), H. S. Joo (Dept. of Mech. and Aerosp. Eng., Seoul National Univ., Seoul, South Korea), S.-J. Shin (Dept. of Mech. and Aerosp. Eng., Inst. of Adv. Aerosp. Technol., Seoul National Univ., Seoul, South Korea), and J. W. Park (Agency for Defense Development, Daejeon, South Korea)

Intense noise produced by the supersonic jet plume from a rocket can cause damage to its structural system and even a failure during launch. In order to predict the acoustic loading by rocket jet noise, large-eddy simulation (LES) is commonly employed for analysis of the jet flow field that is subsequently used as a source condition for predicting the acoustic field. Although desirable from the accuracy standpoint, LES of a supersonic jet is often burdened by an overwhelming computational cost in both runtime and memory. In this talk, we discuss the sensitivity of acoustic predictions on the spatial resolution of LES, and propose a guideline for reducing the mesh requirement. . Here, the flow field of a supersonic jet is calculated using a hybrid RANS/LES scheme on meshes of different sizes, and is fed into the Helmholtz-Kirchhoff integral for prediction of the acoustic farfield. Changes in overall sound pressure level (OASPL) and directivity of the Mach radiation are monitored as the LES mesh varies in size. The study suggests that (a) the accuracy of the acoustic prediction is not much affected by the use of a coarser mesh to a certain extent and (b) the resulting improvement in computation speed and economy thus outweighs the loss of details in LES, given the prediction of acoustic loading as the ultimate goal. (This work was conducted at High-Speed Vehicle Research Center of KAIST with the support of the Defense Acquisition Program Administration and the Agency for Defense Development under Contract UD170018CD.)

10:40

1aNS9. Improvement of aero-vibro acoustic simulation technique for prediction of acoustic loading at lift-off. 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)

Aero-vibro acoustic simulation for the prediction of harmful acoustic loading at lift-off of launch vehicle is developed. In this simulation technique, high-fidelity large-eddy simulation with computational aeroacoutics based on full-Euler equations is employed for computing jet aeroacoustics and their propagation to the outside of payload fairing. Acoustic field inside the payload fairing is computed by the coupled vibro-acoustic simulation based on finite element method. A simplified fairing model is used for the validation of the present method. An impact hammer test and acoustic vibration test using a loudspeaker in an anechoic chamber are conducted for validating the structural model. Then, the accuracy of this method is validated by using the acoustic vibration test result with a subscale rocket engine.

11:00

1aNS10. Spectral analysis of jet turbulence and radiated sound. Oliver T. Schmidt (MAE, Univ. of California, San Diego, 1200 E California Blvd., MC 104-44, Pasadena, CA 91125, oschmidt@caltech.edu), Tim Colonius (MCE, California Inst. of Technol., Pasadena, CA), Aaron Towne (CTR, Stanford Univ., Pasadena, CA), Georgios Rigas (MCE, California Inst. of Technol., Pasadena, CA), and Guillaume A. Brès (Cascade Technologies Inc., Palo Alto, CA)

Informed by LES data and resolvent analysis of the mean flow, we examine the structure of turbulence in jets in the subsonic, transonic, and supersonic regimes. Spectral (frequency-space) proper orthogonal decomposition is used to extract energy spectra and decompose the flow into energy-ranked coherent structures. We demonstrate that two distinct mechanisms, which can be distinguished by their characteristic frequency scaling and spatial support, lead to the formation of wavepackets—coherent structures that are known for their acoustic importance in the aft-angle radiation of high subsonic and supersonic jets. We compare these characteristics to acoustic source features extracted from hologram sound pressure measurements in a recent publication. The evidence strongly suggests that both mechanisms are active in full-scale jets and comprise the experimentally educed sources of sound.

11:20

1aNS11. Launch vehicle acoustic measurements for community noise model development and validation. Alexandria R. Salton, Michael M. James (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, Alex.Salton@blueridgeresearch.com), Matthew F. Calton (Blue Ridge Res. and Consulting, Provo, UT), Kent L. Gee, Reese D. Rasband, Daniel J. Novakovich, and Brent O. Reichman (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A measurement campaign is being conducted under ACRP Project 02-81 to compile a database of high-fidelity modern rocket propulsion noise measurements for facilitating community noise model development and validation. As part of this measurement campaign, acoustic measurements of the Orbital ATK Antares rocket, launched as part of NASA's eighth cargo resupply mission (OA-8E) to the International Space Station, were collected. The OA-8E acoustic measurement test plan was based on guidelines from the community noise measurement protocol being developed under the same effort. Pressure measurements were collected from 46 microphones at 19 unique geographic locations ranging from 0.2 to 19.1 km of the launch pad. The acoustic measurements and the resultant initial data products related to the OA-8E Antares launch will be presented. Comparisons of the multiple recording systems and microphone heights/orientations will be discussed in relation to their relevance in informing equipment recommendations and site layout related to the protocol design.

11:40

1aNS12. Experimental study of acoustic reduction technique 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), Masao Takegoshi (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. Last year the results of the first campaign of the HARE, which aims at understanding the effects of elevation, the shape of ML, were presented. This year, the results of the second campaign which aims at studying the effects of acoustic reduction techniques such as acoustic shields and water injection will be presented.

Session 1aPAa

Physical Acoustics and Signal Processing in Acoustics: Acoustic Metamaterials and Super-Resolution Imaging

Matthew D. Guild, Cochair
Acosutics Div., Naval Research Lab., Code 7165, 4555 Overlook Avenue, SW, Washington, DC 20375

Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Chair's Introduction—8:15

Invited Papers

8:20

1aPAa1. Acoustic imaging with a metamaterial Luneburg lens. Steven Cummer, Yangbo Xie (Dept. of Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, cummer@duke.edu), Yangyang Fu (College of Phys., Optoelectronics and Energy, Soochow Univ., Suzhou, China), Junfei Li, Zhetao Jia, Chen Shen (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC), Yadong Xu (College of Phys., Optoelectronics and Energy, Soochow Univ., Suzhou, China), and Huanyang Chen (Dept. of Electron. Sci., Xiamen Univ., Xiamen, China)

The Luneburg lens is a spherically symmetrical gradient refractive index device with unique imaging properties. Its wide field-of-view and minimal aberration have led it to be successfully applied in microwave antennas. However, only limited realizations of this type of image formation have been demonstrated in acoustics. Here, we apply a new design method for scalable and self-supporting metamaterials to experimentally demonstrate Luneburg lenses for airborne ultrasound. Experimental imaging results are demonstrated.

8:40

1aPAa2. Super-resolution using a microscale membrane array metasurface. Shane W. Lani (Johns Hopkins Appl. Phys. Lab, 11100 Johns Hopkins Rd., M.S. 8-220, Laurel, MD 20723, lani.shane@gatech.edu), Karim G. Sabra, and F. Levent Degertekin (Georgia Inst. of Technol., Atlanta, GA)

A metasurface or 2D metamaterial composed of a membrane array can support an interesting acoustic wave field. These waves are evanescent in the direction normal to the array and propagate in the immersion fluid along the metasurface. These waves are a result of the resonant membranes coupling to the fluid medium and propagate with a group and phase speed lower than that of the bulk waves in the surrounding fluid. Additionally these membranes can generate and detect membrane displacement capacitively such as in an array of Capacitive Micromachined Ultrasonic Transducers (CMUT) as a specific example. A model is developed that can solve for the modes of the membrane array in addition to transiently modeling the behavior of the array to help show the performance of the array in relation to wave propagation. Potential applications of this wave field are examined in the context of subwavelength focusing and imaging. Several methods of acoustic focusing are used on an array consisting of dense grid of membranes and several membranes spatially removed from the structure. Subwavelength acoustic focusing to a resolution of $\lambda/5$, limited by the size of one membrane, is shown in simulations and verified with experiments. This fundamental work in characterizing the waves above the membrane metasurfaces is expected to have impact and implications for transducer design, resonant sensors, 2D acoustic lenses, and subwavelength focusing and imaging.

9:00

1aPAa3. Superresolution through regular sampling: Theory and applications. Annie Cuyt and Wen-shin Lee (Dept. of Mathematics and Comput. Sci., Univ. of Antwerp, Middelheimlaan 1, Antwerp, Antwerpen 2020, Belgium, annie.cuyt@uantwerpen.be)

Estimating fine-scale spectral information from coarse-scale measurements is an important issue in many signal processing applications. The problem of superresolution is therefore receiving considerable attention. We offer a technique that allows to overcome the Shannon-Nyquist sampling rate limitation and at the same time may improve the conditioning of the numerical linear algebra problems involved. The technique is exploiting aliasing rather than avoiding it and maintains a regular sampling scheme [3]. It relies on the concept of what we call an identification shift [1, 2], which is the additional sampling at locations shifted with respect to the original locations in order to overcome any ambiguity in the analysis. Neither the original sampling nor the identification shift need to respect the Shannon-Nyquist sampling theorem. So far the technique shows great potential in magnetic resonance spectroscopy, vibration analysis, echolocation, music signal processing, ISAR radar problems, and DOA or direction of arrival. [1] Annie Cuyt and Wen-shin Lee, "Smart

data sampling and data reconstruction," US patent 9,690,749. [2] Annie Cuyt and Wen-shin Lee, "Smart data sampling and data reconstruction," EP2745404B1. [3] Annie Cuyt and Wen-shin Lee, "How to get high resolution results from sparse and coarsely sampled data," ArXiv e-print 1710.09694 [math.NA].

9:20

1aPAa4. Potential applications of high-index acoustic metamaterials. Farzad Zangeneh Nejad and Romain Fleury (EPFL, EPFL - STI - LWE - ELB 030, Station 11, Lausanne 1015, Switzerland, farzad.zangenehnejad@epfl.ch)

In general, sound propagates in solid materials with a speed that is larger than that of air. Here, we discuss and demonstrate a simple metamaterial design that provides a sound speed slower than that of air. We highlight the potential of high-index acoustic metamaterials for open waveguiding, real-time analog acoustic signal processing, acoustic birefringence, and beam splitting.

9:40

1aPAa5. Far-field superresolution imaging using shaped acoustic vortices. Matthew D. Guild, Jeffrey S. Rogers, Charles Rohde, and Gregory Orris (Acoust. Div., Naval Res. Lab., Code 7165, 4555 Overlook Ave., SW, Washington, DC 20375, matthew.guild@nrl.navy.mil)

The ability to overcome the limitations on resolution due to the effects of diffraction has attracted significant attention in recent years. Previously proposed methods to overcome this limit, and therefore achieve superresolution, have largely been restricted to operating within the near-field region of the aperture. In this work, we will describe how acoustic helicoidal waves can create acoustic vortices that are well below the resolution limit, and how this can enable far-field superresolution acoustic imaging. The acoustic vortices generated in this manner propagate from the near-field into the far-field through an arrangement of stable integer mode vortices, thereby enabling the generation of far-field superresolved features in the acoustic pressure field. In this paper, theoretical and numerical results will be presented for an acoustic aperture which is capable of generating superresolved far-field features in the radiated acoustic pressure, and results will be shown illustrating the superresolution capability of this novel technique. [Work supported by the Office of Naval Research.]

Contributed Paper

10:00

1aPAa6. Noise correlation in a metamaterial: From laboratory to field data. Aida Hejazi Nooghabi (Sorbonne Univ., 4, Pl. Jussieu Case 129, T.46-00, Et.2, Paris 75252, France, aida.hejazi@gmail.com), Julien de Rosny (ESPCI Paris, PSL Res. Univ., Institut Langevin, Paris, France), Lapo Boschi (Sorbonne Univ., Paris, France), and Philippe Roux (ISTERRE, Grenoble Alpes Univ., Grenoble, France)

In the METAFORET experiment, a seismic survey is conducted in a 120 m×120 m flat area, partly occupied by a relatively regular grid of tall pine trees, and partly by a canola field. We study the scattering effects of trees on cross correlations of ambient signal. A wave field is generated by several arrays of sources and recorded at a dense array of receivers within the area of interest. In parallel, we conduct a lab experiment, where the Earth's subsurface and the trees (resonators) are idealized by a thin elastic (aluminum) plate and an array of rods, respectively. The Lamb waves propagating in plate are inherently 2-D and dispersive; thus, a good analogue of seismic surface waves observed in field data. Based on the reciprocity theorem, we can treat receivers as sources and vice-versa, resulting in a virtually uniform source distribution. We auto-correlate recordings corresponding to each virtual source-receiver pair, and visualize the auto-correlation maximum as a function of virtual-source location: this provides a map of energy contributed by virtual sources, inside and outside the region of resonators. From that, we identify the metamaterial position as well as bandgap and propagation band. Results from field and laboratory data are compared.

10:15–10:30 Break

Invited Paper

10:30

1aPAa7. Ultra-compact reflective metasurface for wave manipulation. Jian Chen and Zheng Fan (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., Block N3, Singapore 639798, Singapore, chen_jian@ntu.edu.sg)

Recently, electromagnetic metasurfaces have witnessed a promising future for useful devices with extremely compact footprint. However, their mapping to acoustic regimes remains challenging because the acoustic wavelength is considerably larger than the optics, especially at low frequencies. We propose an ultra-compact metasurface that can manipulate the reflected waves upon plane waves incidences. Remarkably, the metasurface simply consists of a rigid plate perforated with deep-subwavelength grooves of varying depths. A theoretical formulation based on multiple-scattering and dynamical diffraction is established to address the underlying mechanism, and used to optimize metasurfaces for wave manipulation. Numerical simulations and experimental measurements are conducted on the designed sample, which are in good agreement and well predicted by the theory. This study provides a comprehensive guideline for the design of ultra-compact and planar acoustic reflective metasurface; moreover, the study may promise an additional avenue to integrate acoustic devices in practical applications.

Contributed Papers

10:50

1aPAa8. Wavefront shaping of acoustic waves in scattering media.
Yuning Guo (Dept. of Mech. Eng., Univ. of Colorado Boulder, 427 UCB, 1111 Eng. Dr., Boulder, CO 80309, yuning.guo@colorado.edu)

As an essential necessity for fundamental study and a broad variety of technological applications, it is important to control the propagation of acoustic waves and select their particular wavefronts. This work is to achieve effective wavefront shaping of acoustic waves along specific paths in strongly scattering media including phononic periodic structures and random media. In the phononic crystal, a kind of artificial phononic structures, wavefront shaping, and super-resolution imaging have been achieved by designing the geometry and material parameters. Furthermore, the confocal technique of the optical beam and acoustic beam has been improved to achieve sub-wavelength imaging in random media with strongly scattering. The presented results of wavefront shaping have implications for tailoring phonon dynamics in scattering media, which offers further possibilities of controllable acoustic waves in complex materials and enlightens the interaction of multi-physics fields.

11:05

1aPAa9. Inverse design method in acoustic wave manipulation.
He Gao and Jie Zhu (Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong SAR, China, gaohe077@gmail.com)

We present here the involvement of inverse design method with the acoustic wave modulation, to simplify how we get the required material parameters and design novel acoustic devices that can perform certain prescribed operation. As an example, the acoustic cloaking will be discussed. The existing acoustic invisibility design based on coordinate transformation method needs negative index media which can be realized through complicate structures. With inverse design method, we have designed a circular cloak device with positive index materials, that successfully make a circular scatterer acoustically invisible. This inverse design method takes a significant step towards realizing practical cloaking devices and would open up new possibilities for manipulating acoustic wavefront with simplified parameter distribution and optimized properties.

11:20

1aPAa10. A kind of polar coordinate system adaptable metasurface and its broadband wavefront manipulation.
Shanjun Liang (Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong PolyU CF306, Hung Hom, Hong Kong, s.j.liang@connect.polyu.hk), Tuo Liu (Mech. Eng., The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), He Gao (Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Fei CHEN, and Jie Zhu (Mech. Eng., The Hong Kong Polytechnic Univ., Kowloon, Hong Kong)

We report a type of polar coordinate system adaptable metasurface by which a three-dimensional Bessel beam in broadband of 2–6 kHz is realized. For the common design of metasurface, many kinds elements are for a square lattice in the Cartesian coordinate system while sometimes a cylindrical symmetric source can be more suitable for practical working conditions, the 3D Bessel beam for example. Here, we present a unit cell for metasurface by embedding helical blades into an element of required shape. Thus, the lattice structure can be adjusted easily to the distinct coordinate systems. Hence, the arrangement of the elements can be more flexible, which means a larger freedom of design for the requirement of bi-anisotropic metasurfaces can be obtained. With the advantage of helical blades, an ideal control over energy transmission efficiency can be achieved too. Based on this, we investigated the characteristics and performance of the elements and realized a better performed Bessel beam in three dimensional. This design thinking can also be utilized to other kinds of wavefront modulation in broadband for both transmission and reflection control.

11:35

1aPAa11. A rigid acoustic metamaterial with phase modulation and power attenuation.
Nansha Gao (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, gaonansha@nwpu.edu.cn)

We introduced a rigid structure into the acoustic metasurface design, proposed labyrinth structure based on the equivalent medium theory and different media are replaced by curly labyrinth. Layered medium theory and equivalent medium theory are combined to design arbitrary acoustic metasurface structure. An acoustic metasurface studied in this paper and realised simultaneous phase modulation and power attenuation in the air, the effective range covered from 30 to 90 degrees, and power attenuation is over 40%. According to layered medium theory which could modulate acoustic wave direction, the metasurface with same function can also be applied to underwater. Its related simulation results are calculated by FEA method. Finally, by introducing the curly labyrinth theory, the underwater acoustic metasurface with simultaneous phase modulation and power attenuation is designed and verification. This paper potential applications in the rigid underwater acoustic metasurface in low frequency, adjustable direction, and sound attenuation.

Session 1aPAb

Physical Acoustics, Noise, and ASA Committee on Standards: Outdoor Sound Propagation I

Vladimir E. Ostashev, Cochair
U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290

Chair's Introduction—9:00

Invited Papers

9:05

1aPAb1. On the origin of thunder: Reconstruction of lightning flashes, statistical analysis, and modeling. Arthur Lacroix (CEA, Paris, France), Thomas Farges (CEA, Arpajon, France), Régis Marchiano (Institut Jean Le Rond d'Alembert, Sorbonne Université, Paris, France), and François Coulouvrat (Institut Jean Le Rond d'Alembert, CNRS, Université Pierre et Marie Curie, 4 Pl. Jussieu, Paris 75005, France, francois.coulouvrat@upmc.fr)

Thunder from 27 natural lightning flashes of three thunderstorms has been recorded in 2012 in Southern France in the 0.1–180 Hz frequency bandwidth, using a 50 m-wide triangular array of 4 recalibrated microphones in the 0.3–20 km distance range from lightning. Source reconstruction allows to separate, within the acoustical signal, Cloud-to-Ground (CG) from Intra-Cloud (IC) parts of the discharge. The possibility to separate nearby CG events is shown. A total of 36 CG signals and associated spectra is obtained, along with some IC signals. The combination of reconstruction, separation, and frequency analysis provides new insights on the origin of thunder. Thunder infrasound is shown unambiguously to originate dominantly from return strokes. Spectra of CGs and ICs are similar, but of higher amplitude for CGs. No sharp frequency peaks can be clearly evidenced. The influence of distance, therefore of propagation effects, is pointed out. Best fits of energy and frequency gravity center dependence with distance are in agreement with a nonlinear line source propagation. A link between acoustic energy and impulse Charge Moment Change (iCMC) is also indicated. Lightning is modeled as a randomly tortuous line source, and the resulting spectra are compared to observations.

9:25

1aPAb2. Nonlinear reflection of weak shock waves from a rough surface in air. Maria M. Karzova (Phys. Faculty, Moscow State University, Moscow, Russian Federation; Univ. Lyon, CeLyA, Ecole Centrale de Lyon, LMFA - UMR CNRS 5509, Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Thomas Lechat (Univ. Lyon, Ecole Centrale de Lyon and LMFA UMR CNRS 5509, F-69134, Ecully, France), Sébastien Ollivier (Univ. Lyon, Université Lyon 1 and LMFA UMR CNRS 5509, F-69622, Ecully, France), Didier Dragna (Univ. Lyon, Ecole Centrale de Lyon and LMFA UMR CNRS 5509, F-69134, Ecully, France), Petr V. Yuldashev, Vera Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Philippe Blanc-Benon (Univ. Lyon, Ecole Centrale de Lyon and LMFA UMR CNRS 5509, F-69134, Ecully, France)

Irregular reflection of weak acoustic shock waves occurs under the framework of the von Neumann paradox. In this study, the influence of the surface roughness on the reflection pattern was studied experimentally using spark-generated spherically divergent *N*-waves of 1.4 cm length reflecting from rigid rough surfaces in air. Dimensions of the roughness were varied from 20 up to 500 μm for different surfaces. A Mach-Zehnder interferometry method was used to reconstruct the pressure waveforms near the surface. The reconstruction was performed by applying the inverse Abel transform to the phase of the signal measured by the interferometer. It was shown that the height of the Mach stem became shorter for surfaces with larger dimensions of the roughness and disappeared when the surface roughness was large enough. Such tendency was also observed in simulations based on the Euler equations where the acoustic source was introduced as a Gaussian-envelope energy injection and the roughness was either sinusoidal or random and described by a Gaussian correlation function. [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).]

9:45

1aPAb3. Modeling of acoustic pulse propagation and beamforming in forests. Michael B. Muhlestein, Vladimir E. Ostashev, and D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Acoustic pulse propagation through forests is important for many applications, including noise attenuation by stands of trees, and localization of sound sources. Due to the highly complicated distribution of trees in natural forests, it is appropriate to consider a forest as a continuous distribution of scatterers of various shapes and sizes. A propagation model based on radiative transfer theory under a modified Born approximation may be developed for this situation to describe both the coherent and diffuse sound propagation . The simple case of an impulse in an infinite homogeneous forest of diffuse scatterers is first considered, and then the effects of successively including non-diffuse scatterers, ground reflections in a forest of finite height, and, finally, a realistic forest model are analyzed. These theoretical findings are then compared with experimental results. Lastly, a numerical example describing the effect of a forest on a simple beamforming array is considered.

10:05

1aPAb4. Correspondence between sound propagation in discrete and continuous random media. Vladimir E. Ostashev, D. K. Wilson, Michael Muhlestein (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd, Hanover, NH 03755, vladimir.ostashev@colorado.edu), and Keith Attenborough (Open Univ., Leighton Buzzard, Bedfordshire, United Kingdom)

Wave propagation in random media is important in many applications such as sound propagation in a turbulent atmosphere and scattering by bubbles and microparticles in the ocean. Formulations for the statistical moments of the sound pressure field in continuous and discrete random media are usually done independently. In this presentation, it is demonstrated that the equations for the first two statistical moments in a continuous random medium have the same form as those for a discrete random medium if the scattering properties of the media are expressed in terms of the differential scattering cross section and total cross section. This analogy enables us to apply methods developed in wave propagation in continuous random media to discrete media, and vice versa. As an example, the existing theory of the interference of the direct and ground reflected waves in a turbulent atmosphere is used to study the effect of trees on the interference of these waves in a forest. The results obtained are compared with experimental data. The correspondence between wave propagation in discrete and continuous random media can also be used in other fields of physics.

10:25–10:40 Break

10:40

1aPAb5. Bayesian estimation of mean transmission loss along multiple paths with randomly scattered signals. D. K. Wilson (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), and Vladimir E. Ostashev (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A general problem is considered in which a source of unknown power transmits to multiple receiver locations. The signal is randomly scattered along each transmission path, for example, by turbulence, a forest, or buildings. At each receiver location, one or more observations of the signal power are collected. It is assumed that the functional form of the probability density function (pdf) of the received signals (an exponential pdf for strong scattering, or a gamma pdf for weak scattering) is known based on an understanding of the scattering process, although the mean transmission losses (TLs) from the source to receivers are uncertain. From the signal observations, we wish to estimate the mean power of the source and the mean TLs for each path. A Bayesian formulation for this problem is presented. An inverse gamma prior is used for the TL, which is the conjugate prior for the exponential or gamma scattered signal pdf (likelihood function). Analytical solutions are then derived for a number of limiting scenarios: (1) unscattered signals along multiple paths with dependent TLs, (2) strongly scattered signals along multiple paths with the same TL, and (3) weakly or strongly scattered signals along multiple paths with independent TLs.

11:00

1aPAb6. An improved numerical scheme for predicting sound propagation in a vertically stratified medium. Yiming Wang and Kai Ming Li (Mech. Eng., Purdue Univ., 140 South Russell St., West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

Propagation of sound in a stratified medium has been studied for many decades. The ray tracing method, which is one of the most popular ways to solve this kind of problems, involves identifying the sound rays connecting the source with the receiver. However, the ray method does not yield accurate solutions when either the source or the receiver is placed near a turning point. Under these circumstances, it is found necessary to use a wave-based approach leading to a more accurate solution for predicting the sound fields. Typically, the solution is written in a form of a highly oscillatory integral that can be evaluated numerically. High computational times are required to obtain accurate solutions. There have been significant developments in the numerical approximation of highly oscillatory integrals. An improved scheme, which is known as Levin's collocation method, has low computational costs but it gives accurate numerical solutions for high-frequency sound fields. In this paper, the propagation of sound in a stratified and unbounded medium is considered where there is only one turning point. The Levin collocation method is explored for calculating the sound fields efficiently. In addition, an asymptotic solution is also derived for comparison with the Levin collocation method.

11:20

1aPAb7. Nondimensional analysis of wind noise and atmospheric surface-layer properties. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil), Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Christopher M. Hocut (U.S. Army Res. Lab., White Sands Missile Range, NM)

Wind noise is a prominent limitation to the signal to noise ratio of acoustic sensors. Realistic expectations of signal detectability can be generated by predicting the noise floor prior to a sensor deployment; however, a relationship must be established between the wind noise measured by a sensor and atmospheric surface-layer properties. Under conditions of horizontal homogeneity and quasi-steadiness, Monin-Obukhov similarity theory relates friction velocity, temperature scale, and roughness length to the near-surface profiles of mean wind speed and turbulent intensity, which in turn are known to govern wind noise. It is expected that the ratio of one-third octave band root-mean-square sound pressure to the turbulent flux of momentum, Strouhal number, and dimensionless elevation have a nondimensional relationship that collapses wind noise data as a function of Monin-Obukhov parameters. In order to establish such a relationship, we analyze a dataset of wind noise recorded in Spring 2018 within the Army Research Laboratory's Meteorological Sensing Array on the Jornada Experimental Range, New Mexico. This dataset consists of continuous recordings of ambient noise at several sites on audio microphones up to 20 m above ground level, co-located with a suite of high-fidelity meteorological instruments, including sonic anemometers.

11:40

1aPAb8. Calculations of low-frequency wind noise along a low two-dimensional hill surface. Gregory W. Lyons (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, gregory.w.lyons@erdc.dren.mil) and Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Measurement of outdoor sound propagation is often limited by wind noise, i.e. pressure fluctuations from atmospheric turbulence, especially for infrasound and low audible frequencies. Over a flat ground surface, the spectral density of wind noise can be predicted by the shear-turbulence mechanism for static pressure fluctuations using a mirror flow atmospheric turbulence model for the inhomogeneous surface-blocking effect. This study moves beyond a flat ground model to consider the effects of flow distortion by weak topography on surface wind noise, specifically, flow over a low two-dimensional hill, free from separation, at large Froude number. The integral solution for the pressure Poisson equation is used as a starting point, with turbulence modeled by the mirror flow in surface-following coordinates. The perturbation analysis of Hunt *et al.* [Q. J. R. Meteorol. Soc. 114(484), 1435–1470 (1988)] is used to model the mean shear rate as a function of elevation and distance over the hill. For upwind, downwind, and crest positions along the hill surface, the pressure correlation solution is integrated numerically to evaluate one-dimensional spectra. The relative forms of these spectra are analyzed to describe the effects of the hill crest acceleration and downwind wake deficit. Implications for microphone placement are also considered.

Session 1aSAa

Structural Acoustics and Vibration, Physical Acoustics, Underwater Acoustics, and Architectural Acoustics: Advanced Modeling Techniques for Computational Acoustics

Kuangcheng Wu, Cochair
NSWCCD, 9500 MacArthur Blvd., West Bethesda, MD 20817

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg. 1302, Newport, RI 02841

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Invited Papers

8:00

1aSAa1. Scalar metrics for structural acoustic system analysis and differentiation. Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., P.O Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu)

As computational resources continue to grow and as researchers are better able to harness these resources, the size and complexity of numerical models for studying structural acoustic systems will only increase. However, despite these ever larger models, there remains a need to reduce the results into an easily interpreted form. In this work, a variety of scalar metrics—where *scalar metric* is used to refer to any scalar-valued function of either frequency or time—are presented and evaluated for their utility in displaying relevant features of structural acoustic models. Particular attention is paid to metrics that allow an analyst to differentiate between related models, such as when performing a design study where several different options are being considered. The metrics are ranked for several example problems according to how well they describe the features of interest as well as how readily they may be computed.

8:25

1aSAa2. Improving multiple model parameters of a complex fluid-loaded structure from acoustic measurements. Alyssa T. Liem and James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, atliem@bu.edu)

A method for correcting multiple mechanical and acoustical properties in a finite element model using the Neumann series is presented and demonstrated with numerical examples. In previous work, the authors developed a method for estimating a model parameter in a complex structure using a Neumann series as an approximation to system response. The method computed the sensitivity of the system response due to changes in the parameter and subsequently determined the change needed in the parameter to bring model response into agreement with vibration measurements. This previous work demonstrated the accuracy and computational efficiency of the method when correcting one model parameter. In the present work, the authors extend the analysis to compute sensitivities for multiple parameters of a system, allowing for the correction of more than one parameter. The limits and accuracies of the method are explored for a canonical acoustic system in which a complex structure interacts with an acoustic medium. Two uncertain model parameters of the system are corrected by bringing model response into agreement with acoustic measurements. Results of these examples will be reviewed and presented to illustrate the accuracy and robustness of the method.

8:50

1aSAa3. On the study of vibrational interactions of an internal substructure with a main structure submerged in water and its acoustic radiations using admittance approach. Pei-Tai Chen (Dept. of System Eng. and Naval Architecture, National Taiwan Ocean Univ., No.2,Pei-Ning Rd. Keelung 20224, Taiwan, ptchen@mail.ntou.edu.tw)

It is important and practical to design an internal substructure for supporting machines which generates vibration sources for a main submerged stiffened shell structure. Vibration propagates from the internal supporting structure to the wetted shell structure, thus radiating acoustic energy into water. The present study can be divided into two categories: 1. the main wetted structure, including stiffeners and bulkheads, etc., radiates acoustic energy into water subject to forces which is the junction interacting forces arisen from the vibrating machine, 2. the interaction force pertains to the coupling dynamic characteristics between wetted structure and the internal structure. An admittance approach is adopted to characterize individual structures and the coupled equation is derived by continuity of displacement variables and equaling forces with opposite sign at the junction of connected structures. Admittance matrices are computed by using junction forces between the structures whereas the responded displacements at the junctions are the elements of the admittance matrices. The coupled admittance equation is complex symmetric matrix. An eigenvalue analysis is performed to investigate the interaction junction forces, accordingly, the radiation characteristics of the coupled main wetted structure under fluid loading in connection with supporting internal structures.

9:15

1aSAa4. Deterministic and statistical parameter characterization in resonant fluid-structure interaction problems. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, timo.lahivaara@uef.fi), Peter Göransson (Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Stockholm, Sweden), and Jacques Cuenca (Siemens Industry Software, Leuven, Belgium)

This research focuses on developing computational methods to estimate model parameters in resonant fluid-structure interaction problems over a wide frequency range by means of model inversion approaches. The considered problems are widely known to be subjected to local minima, which represent a major challenge in the field of parameter identification. In the proposed method, the frequency spectrum is divided into successive substeps allowing to efficiently guide the estimation towards the global minimum, i.e., the true model parameters. The estimation is performed through two frameworks, namely, the deterministic using gradient-based optimization and Bayesian using Markov chain Monte Carlo method. Proposed numerical examples illustrate the effectiveness and potential of the proposed stepwise scheme to find the global minimum and reduce the overall computational burden.

9:30

1aSAa5. On the use of model truncation to extend the applicability of the finite element method to higher frequencies. Anthony L. Bonomo, Joshua McWaters, and Kuangcheng Wu (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.l.bonomo@navy.mil)

Many structural acoustics problems of interest can be modeled as a vibrating elastic structure situated in and fully coupled to an infinite acoustic fluid domain. To model such problems using the finite element method, techniques have been developed to approximately enforce the Sommerfeld radiation condition at the boundary of the computational domain and prevent spurious boundary reflections from adversely affecting the calculated solution. These techniques include radiation boundary conditions, infinite elements, and perfectly matched layers. It is well known that due to computational constraints, the finite element method is often restricted to relatively low frequencies. However, many of the same techniques that have been used to enforce the Sommerfeld radiation condition can also be used to truncate the computational domain further and allow the finite element method to be used to study higher frequency problems where the structural acoustic response is relatively localized. This talk explores this model truncation application.

9:45

1aSAa6. Acoustic wave scattering from a wave-bearing cavity in a rectangular waveguide. Muhammad Afzal and Hazrat Bilal (Dept. of Mathematics, Capital Univ. of Sci. and Technol., Islamabad 44000, Pakistan, dr.mafzal@cust.edu.pk)

This article discusses the acoustic scattering in a waveguide containing flexible cavity bridged by vertical membranes. A tailored-Galerkin approach is adopted for the solution of governing boundary value problem. Unlike to the usual Mode-Matching (MM) technique, the schemes adopted here for solution modify the MM procedure by using Galerkin and Modal approaches. In Galerkin process, the displacement of vertical membrane is expanded by means of the usual orthogonal modes whilst in the later case, this displacement is found by utilizing the non-orthogonal modes of wave-bearing cavity. The results for scattering powers and transmission loss are shown against frequency and the dimensions of the chamber. A good agreement in results obtained via Galerkin and Modal approaches for different sets of edge conditions is seen. The numerical results show that the choices of edge conditions significantly affect the transmission loss and scattering powers.

10:00

1aSAa7. Multi-layer actuator array to render a vibration field on a point-excited panel speaker. Ki-Ho Lee, Jeong-Guon Ih, and Youngjin Park (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, k.h.lee@kaist.ac.kr)

Panel speakers often adopts the vibration actuator attached to a plate center to excite the whole panel. When a thin rectangular plate is excited by a point force, the generated bending wave is reflected quickly from the edges, so the plate is governed by the reverberant field. Because many vibrational modes participate even in the low frequencies, the radiated sound spectrum is involved with many peaks and troughs resulting a poor sound quality. To minimize the modal participation, the vibration is rendered to be confined and in-phase within a circular area enclosing the actuator point, and the vibration is being suppressed outside of it. An additional multi-layer actuator array surrounding the speaker zone is employed to control the vibration, thus fulfilling the rendered field. The study aim is now to obtain an appropriate gain of the actuators by solving the inverse problem consisting of the transfer matrix between field points and control actuators. The effect of the number of arrays is tested for the radius of 0.05–0.2 m. The control result reveals that the signal-to-noise ratio in the speaker and baffle zone is improved 8–11 dB by the three-layer array than by the single-layer array with the same size.

Session 1aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Signal Processing in Acoustics: Utilization of High-Speed Cameras to Measure Vibration

Micah R. Shepherd, Cochair
Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

Trevor W. Jerome, Cochair
Department of Acoustics, The Pennsylvania State University, GTWT Water Tunnel, State College, PA 16804

Invited Paper

10:45

1aSAb1. Deflectometry, a full field slope measurement technique: General perspectives and application to loading identification using the virtual fields method. Olivier Robin, Patrick O'Donoughue, and Alain Berry (Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Olivier.Robin@USherbrooke.ca)

The ability of photographic techniques to quickly record high volumes of scientific data has been understood since the early 1900s. It is only following the advent of digital imaging and data processing systems that so-called full-field optical measurement techniques became sufficiently reliable for performing high spatial density structural vibration measurements. In particular, the deflectometry technique directly provides a spatially and temporally resolved measurement of slope fields on plane structures using a single high-speed camera. The presentation first demonstrates the principles of this technique and explores some direct applications of such full-field measurements. The advantages and drawbacks of deflectometry are compared with other optical techniques like digital image correlation or scanning laser Doppler vibrometry. A key aspect of several engineering domains consists in the identification of dynamic loads acting on structures by inverse methods. It is shown that coupling deflectometry measurements with the virtual fields method enable the reconstruction of stationary and transient excitations without any specific regularization. Experimental reconstruction results on an aluminum panel are presented for two different transient mechanical loadings: instrumented impact hammer and impacting metal marbles (multiple unknown excitations). Finally, the identification of random excitations is considered using a plate and a membrane.

Contributed Papers

11:10

1aSAb2. Estimating Poisson's ratio of a free, rectangular panel using video-based modal analysis. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu), Olivier Robin (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada), Stephen Hambric (Appl. Res. Lab, Penn State Univ., State College, PA), and Patrick O'Donoughue (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada)

Recent work has shown that the Poisson's ratio of an isotropic material can be determined using the anticlastic curvature that exists in certain mode shapes of a free, rectangular panel of that material. The shapes must be measured experimentally in order to determine the curvature that exists. The curvature is then related to Poisson's ratio based on a relationship that depends on thickness and length-to-width ratio. For accurate determination of the anticlastic curvature, high spatial resolution is required. In this paper, high speed video is used to experimentally measure the mode shapes of a free, rectangular panel. The spatial resolution achieved is much higher than that obtained using traditional methods due to the inherent resolution of the camera. The high-speed video results are demonstrated and compared to modes using traditional modal techniques. The Poisson's ratio is then computed and found to agree well with published values.

11:25

1aSAb3. The problem of parallax when using high speed cameras for measurement. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

When using a high-speed camera as a recording device to measure the displacement or position of an object, the problem of parallax must be properly accounted for, or significant errors in measured positions can occur. In this paper, an experiment to measure the elastic properties of several golf balls of varying construction is described. The goal of the experiment was to verify whether a correlation exists between a ball's coefficient-of-restitution (COR) and the frequency of its lowest structural vibration mode. Balls were dropped from a height onto a rigid surface and the COR was obtained by taking the ratio of speeds just before and just after impact with the rigid surface. Balls were dropped and bounced in front of a scale with fine graduation markings. Video recorded with a high-speed camera was processed using tracking software to measure the ball's position and velocity as a function of time. However, parallax due to the camera field of view introduced significant error into measurement of position and velocity. This paper describes how the parallax error was accounted for in the video-recorded measurements in order to obtain data that matched theoretical expectations.

11:40

1aSAb4. A method for measuring the dynamic parameters by extracting the incident. Hong Hou, Zhengyu Wei, Nansha Gao (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an Shaanxi 710072, China, houhong@nwpu.edu.cn), and Jianhua Yang (School of Automation, School of Automation, Northwestern PolyTech. Univ., Xi'an, China)

The complex Young's modulus of viscoelastic materials can be determined by monitoring the propagation of a traveling burst in a thin bar, which is called wave-speed method. While the test samples are always too short to make the incident wave for lower-frequency signals be detected, which lead to low-frequency test is difficult using this method. To solve this problem, a method of extracting the incident wave is presented in this paper. A pulse wave is generated by the exciter and is used to force the longitudinal vibration of a viscoelastic thin bar. The velocities of the two ends of the thin bar are measured by two laser Doppler vibrometers. The incident waves at both of the two ends of the sample can be easily extracted, and the dynamic parameters of the sample can be obtained by the ratio of the vibration velocities of them. This method expands the experimental frequency range of the wave-speed method and the measured results agree well with that of commercial viscoelastic instrument.

MONDAY MORNING, 5 NOVEMBER 2018 SHAUGHNESSY (FE), 7:55 A.M. TO 12:00 NOON

Session 1aSP

Signal Processing in Acoustics, Acoustical Oceanography, Architectural Acoustics, Musical Acoustics, Underwater Acoustics, and Noise: Machine Learning for Acoustic Applications I

Peter Gerstoft, Cochair
SIO Marine Phys Lab MC0238, Univ of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238

Weichang Li, Cochair
Aramco Research Center - Houston, 16300 Park Row, Houston, TX 77084

Chair's Introduction—7:55

Invited Papers

8:00

1aSP1. Machine learning in acoustic applications with wave physics models, a tutorial. Weichang Li (Aramco Res. Ctr. - Houston, 16300 Park Row, Houston, TX 77084, lwc@alum.mit.edu)

Through recent decades of intensive research, machine learning has established its great potential in areas such as social networking, e-commerce, computer vision, natural language processing, and robotics. Recently, it has also started to capture the attention of the acoustic research community where the problems have wave physics basis and the research has focused on related areas such as signal/image processing. This is evident in the significantly increased number of related papers presented at recent ASA meetings. The challenge appears to concerning several aspects: i) What types of acoustic applications can be formulated as ML problems in a way that could provide interesting results and potentially better performance? ii) What would be the relationship between wave physics models and machine learning methods? iii) What is the dataset requirement of machine learning that is potentially different from say, signal processing techniques, and what implications it might have on experimental design, data collection, and annotation? ii) What are the areas that might not have much benefit from taking a machine learning approach? This tutorial will present a number of acoustics/geophysics application examples along with their analogs in canonical machine learning applications, as an attempt to illustrate possibilities and limitations.

8:40

1aSP2. InversionNet: A real-time and accurate full waveform inversion with convolutional neural network. Youzuo Lin and Yue Wu (Geophys., Los Alamos National Lab., Bikini Atoll Rd., Los Alamos, NM 87545, ylin@lanl.gov)

With data proliferation in all geosciences domains, machine learning and data analytics are emerging as important research areas in geosciences. Full-waveform inversion has been an important tool to infer the subsurface based on geophysical measurements. However, solving full-waveform inversion can be challenging. The existing computational methods for solving acoustic full-waveform inversion are not only computationally expensive but also yields low-resolution results because of the ill-posedness and cycle skipping issues of full-waveform inversion. To resolve those issues, we employ machine-learning techniques to solve full-waveform inversion in this work. In particular, we build a convolutional neural network to model the correspondence from data to velocity structures. Our numerical examples using synthetic reflection data show that our new methods much improve the accuracy of the velocity inversion. Furthermore, the computational time for inverting the seismic data is significantly reduced.

9:00

1aSP3. The machine learning aspects in building the global smartphone seismic network. Qingkai Kong and Richard Allen (Earth & Planetary Sci., UC Berkeley, 289 McCone Hall, UC Berkeley, Berkeley, CA 94720, kongqk@berkeley.edu)

MyShake is a global crowdsourcing smartphone seismic network to monitor and detect earthquakes. After it got released to the public in 2016, we arrived at more than 300,000 downloads with more than 800 detected earthquakes globally within 2 years. Machine learning plays a critical role in MyShake that makes everything happen. In this presentation, I will present the details of how we use the artificial neural network to distinguish earthquakes from the human activity movements recorded on a single phone in real-time. This includes how we do the data acquisition, pre-processing the data, addressing imbalanced datasets, feature engineering/selection, and evaluating the model. I will also talk other machine learning aspects in the MyShake network including the convolutional neural network that we built on the server to further classify the whole waveforms to find that caused by earthquakes, the adversarial machine learning for securities of the system, dealing with the dynamically changing network, training customized model for each user etc. These machine-learning applications in MyShake illustrate the power of combining data science and geophysics and provide good examples of how do we better facilitate the interactions of the two fields.

Contributed Papers

9:20

1aSP4. Total independent energy distance—A measure for choosing efficient spectrogram resolutions. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

This paper describes a quantitative method for choosing the most efficient frequency resolution and window overlap when creating spectrograms. The method calculates the Euclidean distance, d, and the coefficient of determination, R^2, between a pair of successive power spectra in the spectrogram. The quantity $d \times (1—R^2)$ is the Euclidean distance between the windows' power spectra, discounted by the information redundancy between the spectra. The Total Independent Energy Distance (TIED) is then the sum of those distances along the entire spectrogram. The effectiveness of TIED is computed for a variety of artificial signals including white noise, sawtooth waveforms that vary in their frequency resolution, pulse trains, and amplitude-modulated signals that vary in their time distribution, and frequency sweeps that vary in both. The TIED is then calculated for human speech, bird song, and insect buzzes, demonstrating that it provides quantitative support to time- and frequency-resolutions for the spectrograms that closely adhere to the traditional resolutions that have been arrived at through generations of human pattern recognition.

9:35

1aSP5. Dereverberation binaural source separation using deep learning. Dhany Arifianto and Mifta N. Farid (Dept. of Eng. Phys., Insitut Teknologi Sepuluh Nopember,Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id)

This paper reported a deep-learning based binaural separation using gammatone-frequency cepstral coefficient (GFCC) and multi-resolution cochleagram (MRCG) as spectral features and interaural time difference (ITD) and interaural level difference (ILD) as spatial features. A binary mask was estimated by deep neural network (DNN) binary classifier that used the features as a training data and ideal ratio mask (IRM) as a training target. In the experiment, a male speaker as a target speech at azimuth 0^o and a female speaker as a masker speech at azimuth 30^o, 20^o, 10^o, and 5^o in rooms with 0.32, 0.47, 0.68, and 0.89 s reverberation time (RT_{60}). As a training process, 50 mixtures were used for each condition experiment. The classifier contained two hidden layers, 200 binary neurons and 50 epoch, and Restricted Boltzmann Machine (RBM) was used as pre-training process. The RBM and the classifier learning rate was 1 up to 0.001 from epoch 1 to epoch 50. The sound quality results indicated by 88% intelligibility of STOI method which means that the separated sound is easily understood by the listener. The MOS value is 2.8 which means the sentence is clear but spectral distortion is slightly annoying.

9:50

1aSP6. Predicting transmission loss errors through machine learning. Jennifer Cooper, C. J. Della Porta, and Olivia Ott (Johns Hopkins Univ. Appl. Phys. Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu)

Uncertainty in a sound speed profile can lead to uncertainty in the associated prediction of transmission loss (TL). In order to better quantify the effect of imperfect knowledge of the sound speed profile on the acoustic propagation, a study was done comparing pairs of sound speed profiles. In each pair, one profile was treated as ground truth and the second profile was a perturbed version of the first. No single metric on the sound speed profiles, such as mixed layer depth or surface layer characteristics, correlated well with the errors in resulting TL. Several attempts at creating a more complex metric on the profile that could predict errors in the TL were also unsuccessful. However, even a rather simple machine learning approach was able to reliably predict TL errors. Results of the study will be presented and implications discussed.

10:05–10:20 Break

Invited Papers

10:20

1aSP7. Deep learning for estimating porous material properties from full-waveform data. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, timo.lahivaara@uef.fi)

This research focuses on both developing computational tools to model transient wave propagation in coupled poroviscoelastic-viscoelastic-acoustic media and estimating material properties from the recorded full-waveform data. Numerical simulation of wave-dominated problems is computationally demanding. Efficient parallelization, capability to handle complex geometries, and sufficient numerical accuracy are some of the requirements for a suitable full-waveform simulation technique. On the other hand, the robustness and prediction accuracy are needed from the method used to solve the corresponding inverse problem. In this work, the discontinuous

Galerkin method is used to solve the forward model while the convolutional neural networks is used to solve the estimation problem. Two-dimensional model problems with simulated data are presented. In the numerical experiments, the primary unknowns are estimated while the remaining parameters which are of less interest are successfully marginalized.

10:40

1aSP8. Machine learning for behavior classification of passively tracked vessels. Erin M. Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu), Oscar A. Viquez, and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

Situational awareness is a constant challenge for autonomous underwater vehicles (AUVs) due to limited communication to the surface, navigation drift, and a need to operate in busy areas around other vessels. One piece of information that would be valuable to both keep AUVs safe and inform autonomous monitoring missions is the location and behavior of nearby surface vessels. Passive acoustic data collected on a hydrophone array and processed on an on-board computer can provide bearing and time-to-intercept (TTI). This information can be used to classify overall boat behavior and inform AUV response: for example, investigating an area a boat has circled or avoiding an approaching vessel for vehicle safety reasons. Simulation studies were used to characterize trajectories for simple vehicle behaviors based on the bearing and TTI data produced by an existing passive tracking system. A classifier based on K-nearest-neighbor with dynamic time warping as a distance metric was used to classify simulation data. The simulation-based classifier was applied to classify experimental tracking data on boats completing different types of behaviors using bearing/TTI from dock-based and AUV-based hydrophone arrays.

Contributed Papers

11:00

1aSP9. Proof of concept: Machine learning based filling level estimation for bulk solid silos. Paaranan Sivasothy, Matthias Andres, and Gregor Corbin (Dept. of Mathematics, Technomathematics, Gottlieb-Daimler-Straße 44, Kaiserslautern 67663, Germany, sivasothy@mv.uni-kl.de)

Rigid silos are often under pressure and filled with hazardous materials. Therefore, in very few cases their level can be checked visually. For example, the level of mobile bulk solids silos, which are often used on construction sites, is checked by a worker throwing a stone against the silo and uses the sound to estimate the invisible silo cavity. This method is subjective and shows great errors based on experience. The proposed method should implement this principle robustly by machine application. A sensor unit is to give a mechanical impulse to the outer silo wall with a percussion mechanism and record the resulting sound with a microphone. Just as a human being is able to make an experience-based statement about the filling level of the silo over the growing amount of sounds heard, different machine learning approaches are to be considered in order to imitate this procedure. Machine learning approaches usually aim to find patterns between different sizes. A strongly approximated mathematical model will be used to prove that there is a noisy but demonstrable direct correlation between the level and the acoustic impulse response. This should justify future efforts to find suitable machine learning methods for this application.

11:15

1aSP10. Grating lobe prediction and deconvolution for synthetic aperture sonar. Jeremy Dillon (Kraken Robotics, 430 Water St., Ste. 100, St. John's, ON A1C 1E2, Canada, jdillon@krakenrobotics.com)

Synthetic aperture sonar (SAS) arrays are typically undersampled in the sense that the array element spacing is much larger than the acoustic wavelength. Grating lobes are suppressed by making a judicious choice of beampattern nulls for the transmit and receive elements, such as a 3:2 ratio for the length of the transmit and receive elements. However, grating lobe artifacts can appear in SAS imagery when the target strength difference between a highly reflective object and the surrounding seabed exceeds the sidelobe level of the synthetic array. We present a theoretical model of the SAS point scattering function (PSF) that takes into account shaded element beampatterns for a multichannel SAS array. The PSF model is validated using experimental data from AquaPix, a wideband 300 kHz interferometric SAS. In practice, observed SAS images are described by the convolution of the seabed reflectivity with the PSF. Therefore, knowledge of the PSF facilitates the removal of grating lobe artifacts using deconvolution techniques such as the Richardson-Lucy algorithm. Conventional and deconvolved SAS images of a highly reflective target are presented to demonstrate the effectiveness of deconvolution based on the modeled PSF.

11:30

1aSP11. Geospatial estimation of noise levels between sparsely distributed sensor nodes using machine learning. Matthew G. Blevins and Gordon M. Ochi (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, matthew.g.blevins@usace.army.mil)

Monitoring noise levels over large areas is typically limited by the number of sensor nodes. While previous studies have been able to accurately estimate noise levels using densely distributed sensors in confined spaces, such as along roads in urban areas, estimating noise levels with relatively few sensors spaced over a large area remains a challenging problem. Point sampling of noise levels due to single noise events is also limited by the models used to estimate between sensor locations and their inherent assumptions. To address these limitations we propose nonlinear, adaptive, and robust tools to estimate levels between sensor nodes based on machine learning. Random forests, support vector machines, and Gaussian processes are explored along with conventional geostatistical methods such as ordinary kriging. These methods are trained and evaluated on data from both measurement and simulation of blast noise on military testing and training ranges. The performance of the methods is evaluated using cross validation and root-mean-square-error.

11:45

1aSP12. Measurement matrix design for sparse spatial spectrum estimation in Khatri-Rao subspace. Jiajun Shen and Tian Zhou (Harbin Eng. Univ., Rm. 821, underwater Acoust. Bldg., No. 145, Nantong Ave., Nangang District, Harbin, Heilongjiang 150001, China, shenjiajun@hrbeu.edu.cn)

Compressive (or namely compressed) sensing (CS) exhibits superior performance in sparse spatial spectrum estimation from a predefined vandermonde based dictionary. The CS theory requires that dictionary is as incoherent (orthogonal) as possible since the number of atoms is generally much more than observation vectors, and, namely, the dictionary is over-complete or redundant. Previous researches focus on designing sensing matrix to reduce the mutual coherence of dictionary. However, according to Grassmannian frames, it is still a problem that the coherence of a given dictionary is hard to break through an equiangular tight frame (ETF). To address the problem, we proposed and proved a KR-KSVD method to break through the original lower bound of mutual coherence. In the Khatri-Rao subspace, measurement matrix is designed by minimizing the cost function between the Gram matrix of the equivalent dictionary and an identity matrix with the KSVD method. Simulations demonstrate that the method can produce a better performance in terms of mutual coherence property and sparse recovery accuracy.

Session 1aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Variability in Shallow Water Propagation and Reverberation I

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

David R. Dall'Osto, Cochair
Acoustics, Applied Physics Laboratory at University of Washington, 1013 N 40th St., Seattle, WA 98105

Invited Papers

8:00

1aUW1. Assessment of the temporal and spatial dependence of reverberation mechanisms for KOREX-17. Dajun Tang, Brian T. Hefner, and Taebo Shim (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Mid-frequency reverberation data were obtained over a 9-day period off Geoje Island, Republic of Korea, complemented by transmission loss and backscatter measurements. The reverberation data were collected using the Autonomous Reverberation Measurement System (ARMS), a benthic lander with a source and receive array mounted on a rotation stage in order to collect reverberation data as a function of bearing angle. The ARMS was repeatedly deployed in roughly the same location over the course of the experiment. The surficial sediments close to the ARMS are known to be composed primarily of mud with occasional rock outcrops. Beyond a range of 2 km, the sediment composition is uncertain and a geoacoustic measurement survey has been planned to collect additional data in this region. Using contemporaneous measurements of the sea surface directional wave spectrum and the currently limited geoacoustic data, modeling is used to estimate the relative importance of sea bottom and surface reverberation. [Work supported by of the Office of Naval Research.]

8:20

1aUW2. Vector intensity properties from direct and reverberant field from a mid-frequency sonar in shallow water. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu) and David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Intensity Vector Autonomous Recorder (IVAR) is a benthic lander system that coherently measures 3-axis acoustic particle velocity and pressure (combined sensor). Results using IVAR in a mid-frequency propagation and reverberation study conducted near Geoje Island, Republic of Korea, are presented. The measurements were made within shallow semi-circular bay (depth ~25 m) with bay opening to deeper waters, and with directional wave measurements made within the bay waters. This presentation will focus on measurements made at a fixed range (100–1000 m, depending on test) from the Autonomous Reverberation Measurement System (ARMS), a benthic lander with a source and receive array mounted on a rotation stage in order to collect reverberation data as a function of bearing angle. Several vector and scaler metrics emerge based on different combinations of second-order acoustic fields are discussed, such as rate of energy transport and active and reactive intensity that lend insight into the process of shallow water reverberation. For example, in one continuous measurement made over a 12 hour period, there is high correlation between wave slope and horizontal beam width as determined with horizontal active intensity. This and other effects related to variability in shallow water reverberation will be discussed.

8:40

1aUW3. Measurements and modeling of mid-frequency propagation loss during the Korea Reverberation Experiment 2017 (KOREX-17). Su-Uk Son (The 6th R&D Inst., Agency for Defense Development, Jinhae P.O. Box 18, Chanwon, Gyeongnam 51678, South Korea, suson@add.re.kr), Hyuckjong Kwon, Jee Woong Choi (Dept. of Marine Sci. and Convergent Technol., College of Sci. and Technol., Hanyang Univ., Sangnok-gu, Ansan-si, Gyeonggi-do, South Korea), Youngnam Na, Joung-Soo Park (The 6th R&D Inst., Agency for Defense Development, Changwon, Gyeongnam, South Korea), and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Sound propagation in shallow water is significantly influenced by fluctuation of the water medium and scattering from rough ocean boundaries. To study the effect of the intensity fluctuations caused by the various environmental effects, mid-frequency propagation loss measurements along with the ocean environmental measurements were conducted on May 25–31, 2017, in shallow water off Geoje island, as part of the Korea Reverberation Experiment 2017 (KOREX-17). Continuous wave and linear frequency modulated signal with a center frequency of 3.5 kHz were transmitted by the Autonomous Reverberation Measurement System (ARMS). The Self Recording Hydrophone (SRH) was towed by the R/V Mirae at a speed of approximately 3 knots along two different tracks. Sound speed profiles were measured using the moored CTD chain near the source location, covering the water column between 3 and 50 m. CTD casts were also conducted at the beginning and end of each track. In this talk, the fluctuations of measured propagation losses were presented for both tracks. Finally, the effects of the sound speed variations on the propagation loss will be discussed in comparison with the model predictions obtained using the measured sound speed profiles.

9:00

1aUW4. Oceanographic effects on mid-frequency acoustics during KOREX-17. Taebo Shim, Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, tbshim@apl.washington.edu), Su-Uk Son (Agency for Defense Development, Chanwon, Gyeongnam, South Korea), Youngnam Na (Agency for Defense Development, Changwon, Gyeongnam, South Korea), and Dajun Tang (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

A 9-day mid-frequency, shallow water experiment was conducted off Geoje Island, Republic of Korea, in May 2017. The experiment consisted of transmission, reverberation, and backscatter measurements. The experiment site includes a shallow bay, with water depth less than 30 m, which opens to the Korea Strait where the depth reaches 60 m at a range of a few kilometers from the bay entrance. While the bathymetry of the site is well documented, the geo-acoustic properties of the area is complex, comprised of mud with rock outcrops and regions of sand. The oceanography during the experiment was dominated by tidal forcing and this is expected to be the main source of temporal variability in propagation and reverberation at the site. This paper focus on understanding the variability of the water column in space and time by analyzing data from CTD casts and from a CTD chain, supplemented by data from a nearby oceanographic buoy. A preliminary assessment of the impact of this variability on transmission loss is also examine. [Work supported by the Office of Naval Research and the Agency for Defense Development.]

9:15

1aUW5. Geoacoustic inversion of 3.5-kHz towed receiver data from the KOREX-17. Hyuckjong Kwon, Jee Woong Choi (Dept. of Marine Sci. and Convergent Technol., College of Sci. and Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, hjkwon1014@gmail.com), Su-Uk Son (Agency for Defense Development, Chanwon, Gyeongnam, South Korea), Youngnam Na (Agency for Defense Development, Changwon, Gyeongnam, South Korea), Seom-Kyu Jung (Korea Inst. of Ocean Sci. & Technol., Busan, South Korea), Dajun Tang, and Taebo Shim (Univ. of Washington, Seattle, WA)

The Korea Reverberation Experiment 2017 (KOREX-17) was conducted in a shallow water located in the south of the Geoje island, Korea. During the experiment, sound propagation measurements were made using a 3.5-kHz CW signal along different 2 tracks to a distance of ~10 km from a bottom-mounted source, ARMS (Autonomous Reverberation Measurement System). The signals were received by a SRH (Self Recording Hydrophone), which was towed at a depth of ~20 m during the measurements. The sound speed profile was almost iso-velocity, and the sediment at the site was mainly composed of silt having a mean grain size of 6 phi. Since the sound propagation in shallow water is greatly influenced by sound interaction with the sediment, geoacoustic inversion was tried using a genetic algorithm based matched field processing in which the measured acoustic pressure field was compared to the simulated field predicted by a parabolic-equation based propagation model (RAM). The results are compared to the geoacoustic parameters obtained by the empirical relationship to mean grain size and a sediment layering information obtained by a chirp sonar survey. [Work supported by Agency for Defense Development, Korea (UD170014DD).]

9:30

1aUW6. Measurements of mid-frequency bottom backscattering during KOREX-17. Dong-Gyun Han, Raegeun Oh, Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan 15588, South Korea, dghandg@hanyang.ac.kr), Dajun Tang, and Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Measurements of mid-frequency bottom backscattering strength were made on May 30, 2017, as part of the Korea Reverberation Experiment (KOREX-17) in a shallow water region located at the south coast of Korea. The acoustic data were transmitted and received as functions of frequency (4, 6, and 8 kHz) and pulse length (2 and 4 ms) using semi-monostatic sonar system composed of omni-directional acoustic source and receiver, which was deployed from the *R/V* Mirae. The bottom backscattering strength were extracted from intensity-averaged reverberation level for 30 individual pings. The water depth measured by an echo sounder was about 35 m and bottom was approximately flat, having a slope less than 1° over the experimental area. The surficial sediment was estimated to be silty sediment, having a mean grain size of 6 phi from grain size analysis. In this talk, the measurement results of bottom backscattering strengths will be presented as a function of grazing angle and compared to the predictions obtained by Lambert's law and APL-UW scattering model. [Work supported by Agency for Defense Development, Korea(UD160006DD).]

9:45

1aUW7. Rocky outcrops as clutter in mid-frequency reverberation measurements. Brian T. Hefner, Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan, South Korea), and Taebo Shim (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Scattering by isolated rock outcrops stand out as prominent features in mid-frequency reverberation measured during KOREX-17, a shallow water experiment conducted off Geoje Island, Republic of Korea, in May 2017. The reverberation data were collected using the Autonomous Reverberation Measurement System (ARMS), a benthic lander with a directional source and receive array mounted on a rotation stage. This stationary system was deployed in roughly the same position on the seafloor over the course of the 9-day experiment. A side-scan sonar survey of the seafloor was conducted to identify the location and rough spatial extend of the exposed portions of the rock outcrops. This talk examines scattering by the rock outcrops as a function of time, signal waveform, and changing oceanographic conditions, with an eye toward detection in clutter environments. Since the ARMS is a fully-calibrated sonar system, with known source level and directivities, the target strengths of the outcrops are also estimated with the long-term goal of developing a scattering model of the most prominent rock outcrop which extends 7-8 m above the seafloor. [Work supported by ONR.]

10:00–10:15 Break

10:15

1aUW8. Analysis of sonar clutter using backscattered spatial correlation. Chad M. Smith, Daniel C. Brown, and John R. Preston (Appl. Res. Lab., The Penn State Univ., State College, PA 16804, chad.smith@psu.edu)

Sonar clutter events are one of the primary limitations of long-range active sonar systems within the littoral environment. For this reason, research towards understanding their acoustic return characteristics and ultimately their mitigation are of high importance. This paper discusses measurements and modeling of the backscattered horizontal correlation function from clutter events and comparison with returns from the littoral environment. A statistical model for returns from boundaries, as well as general expectations of returns from common clutter sources will be discussed. The horizontal correlation function is estimated via sub-aperture beamforming of the Five Octave Research Array (FORA), a towed line array system owned and operated by Penn State and funded by the Office of Naval Research. [Work supported by Office of Naval Research.]

10:30

1aUW9. Variability in surface reverberation observed with a high-frequency obstacle avoidance sonar. Mark V. Trevorrow (DRDC Atlantic, 9 Grove St., PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, mark.trevorrow@drdc-rddc.gc.ca)

Sea-surface reverberation and, in particular, near-surface bubble backscatter, can pose a strong limitation on HF sonar performance. Recent high-frequency obstacle-avoidance sonar measurements showed significant variability in low grazing angle sea-surface reverberation. The sonar used in these sea-trials operated at 90kHz covering a 90-degree angular sector with 128 beams at ranges up to 600 m. It was projected horizontally forward from a ship at a depth of 3.5 m in deep water. The statistics of the background reverberation under Sea-State 3 to 4 conditions were investigated. The time- and spatially averaged background reverberation levels agreed with well-known APL-UW models. However, the instantaneous reverberation amplitudes exhibited non-Rayleigh statistical distributions, in better agreement with log-normal distributions. This variability was attributed to scattering patchiness and surface wave effects.

10:45

1aUW10. Dynamic imaging of a gravity wave caused by laser-induced breakdown in a fluid waveguide using multi-reverberated ultrasonic waves. Tobias van Baarsel, Philippe Roux (Université Grenoble-Alpes, Université Grenoble Alpes ISTerre, Grenoble 38000, France, tobias.van-baarsel@univ-grenoble-alpes.fr), Barbara Nicolas (Créatis, Villeurbanne Cedex, France), JEROME I. MARS (GIPSA-Lab, Grenoble, France), Michel Arrigoni (ENSTA, Brest, France), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), and Steven Kerampran (ENSTA, Brest, France)

The dynamic imaging of a gravity wave propagating at the air-water interface is a complex task that requires the sampling of every point at this interface during the gravity wave propagation. Using two source-receiver vertical arrays facing each other in a shallow water environment, we manage to isolate and identify each multi-reverberated eigenbeam that interacts with the air-water interface. The travel-time and amplitude variations of each eigenbeam are then measured during the crossing of the gravity wave. In this work, we present an ultrasonic experiment in a 1 m-long, 5 cm-deep waveguide at the laboratory scale. The waveguide transfer matrix is recorded 100 times per second at a sample rate of 1.1 MHz between two source-receiver arrays while a low-amplitude gravity wave is generated by a laser-induced breakdown at the middle of the waveguide above the water surface. The controlled and therefore repeatable breakdown causes a blast wave that interacts with the air-water interface and penetrates into the water, creating ripples at the surface that propagate in both directions. The surface deformation induced by these two wave packets is also measured by two cameras which allows for independent validation of the ultrasonic inversion. The ultrasonic inversion performed from a few thousand eigenbeams lead to accurate quantitative imaging of the dynamic of the air-water interface, using either the travel-time or the amplitude variation of the ultrasonic arrivals.

11:00

1aUW11. Validity of the frozen-surface approximation for large time-bandwidth signals. Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Douglas A. Abraham (CausaSci LLC, Ellicott City, MD), Stefan Murphy (DRDC, Dartmouth, NS, Canada), and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

The physical modeling of underwater acoustic propagation, scattering, and reflection, and the signal processing associated with these processes, usually rely on the frozen-surface approximation. This longstanding approximation, which dates back to Carl Eckart's seminal paper (The Scattering of Sound from the Sea Surface, 1953) assumes that the ocean surface can be modeled as being frozen in time throughout the entire pulse duration. The approximation is valid for short-duration pulses typically used by sonars in the decades following that paper; however, its applicability to present-day high duty cycle sonars is questionable. Although the assumption is based on the physics of the problem, it can have a profound effect on the signal processing, especially for large time-bandwidth signals. In this paper some experimental results will be presented to provide examples of how the approximation fails for a long duration, wideband pulse. Using this as motivation, the authors will explore some areas where incorrectly employing the approximation can introduce errors in the expected signal processing gain. While no attempt will be made to correct shortcomings in the approximation, it is hoped that the discussion may motivate renewed interest in this issue. Funding from ONR and ONR Global is gratefully acknowledged.

11:15

1aUW12. Modeling acoustic interactions with a dynamic rough sea surface boundary. Alex Higgins, Martin Siderius (Elec. & Comput. Eng., Portland State Univ., 1900 SW 14th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu), Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., Halifax, NS, Canada), and Douglas Abraham (CausaSci LLC, Ellicott City, MD)

The acoustic wave scattering properties of a dynamic pressure-release surface boundary are analyzed using a numeric technique based on the finite-difference time-domain (FDTD) method. Of primary interest is to study the impact of assuming a "frozen" sea-surface on long duration sonar transmissions. Although relatively uncommon in ocean acoustics, the FDTD approach is well suited for modeling boundary roughness and motion. This technique has the additional benefit that the pressure fields are resolved over time, which allows for transient analysis of the observed wave scattering effects. The method is adapted from electromagnetic wave scattering and can properly model the physics of the observed system, which shows a frequency modulated reflection that includes a double-Doppler effect. First, a traditional analytic solution for the static smooth surface boundary ocean half-space model, the Lloyd Mirror, is compared to an equivalent simulation using the FDTD method. Then a dynamic smooth surface boundary is investigated using a modified Lloyd-mirror solution and the FDTD method. Finally, surface roughness for both static and dynamic boundary cases will be considered. Agreement is shown between FDTD simulations and the modified Lloyd Mirror model for both one-dimension and two-dimension cases. [Work supported by the Office of Naval Research.]

11:30

1aUW13. Examination of extensions to transport theory that includes rough surface scattering. Eric I. Thorsos and Frank Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu)

Sea surface forward scattering has important effects on shallow water propagation and reverberation at mid frequencies (i.e., 1–3 kHz) under typical sea surface roughness conditions. Coupled-mode or rough surface PE modeling of these effects require averaging results over many rough surface realizations, increasing the computational effort. An alternative method is based on transport theory, where equations are developed for propagating the moments of the field, avoiding the need for utilizing rough surface realizations. Our transport theory method is based on expanding the field in unperturbed modes, and the equations of motion are for moments of the mode amplitudes. The approach has been based on keeping terms to only first-order in the surface height h(x), making the method linear in surface height. Methods for extending the approach beyond the linear model will be described, both with using realizations with coupled modes and with attempts to extend these approaches to obtain a transport theory. A key part of the approach is the use of the Differential Algebraic Equation (DAE) method in which the range-derivative of the effective boundary condition on the mean plane of the rough surface is used instead of the effective boundary condition itself. [Work supported by ONR Ocean Acoustics.]

11:45

1aUW14. Statistical optimal filtering method for acoustical signal deblurring. Viktoria Taroudaki (Dept. of Mathematics, Eastern Washington Univ., 316 Kingston Hall, Cheney, WA 99004, vtaroudaki@ewu.edu), Michael Taroudakis, and Costas Smaragdakis (Mathematics and Appl. Mathematics and IACM, Univ. of Crete and FORTH, Heraklion, Greece)

Acoustical signals in applications of acoustical oceanography, such as ocean acoustic tomography and sea-bed classification using acoustic signals emitted from known sources, are optimally exploited if they are noise free. The effect of blur in acoustic signals has not been well studied, although the blurring mechanism might introduce severe problems in the use of the acoustic signals for specific applications, especially those using the full signal as the carrier of the relevant information. Deblurring of the signals in addition to denoising is therefore essential for the effective use of the signals. In our work, we apply a Statistical Optimal Filtering method that uses the singular value decomposition of a first estimate of the blurring matrix and statistics to deblur the signal in an efficient and effective way and to quantify uncertainty for the recovered signal. In this talk, we will present the method and discuss its effectiveness using as test case, an application of sea-bed classification based on a statistical characterization of an acoustic signal. The statistical characterization is particularly sensitive to noise and blur contamination of the exploitable signal and any attempt for getting a signal clear from noise and blur is absolutely necessary, for obtaining reliable results.

MONDAY AFTERNOON, 5 NOVEMBER 2018 THEATER (VCC), 1:00 P.M. TO 4:20 P.M.

Session 1pAA

Architectural Acoustics and ASA Committee on Standards: Auditorium Acoustics and Architectural Design: Challenges and Solutions 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. Implementing acoustical designs—Non-acoustical project choices may help or hinder successful outcomes. Damian Doria (Stages Consultants LLC, 75 Feather Ln, Guilford, CT 06437-4907, damianjdoria@gmail.com) and Robert Essert (Sound Space Vision, London, United Kingdom)

The authors contributed acoustical design for many auditoria over the last thirty years. Many of those designs incorporated novel and sometimes unconventional approaches to achieving room acoustics goals. Aside from technical demands, implementing new acoustical ideas requires navigating a number of other challenges from discovering and coordinating the needs of project stakeholders through oversight of the construction and commissioning process. Consultants are called to serve as counselors in addition to our role as technical experts and designers. Much depends on the Owner's aspirations, leadership and structure, and on procurement and delivery approaches. A number of case-examples will be presented along with discussion of the sometimes unexpected challenges faced by project teams in successful realization of their designs.

1:25

1pAA2. Case study: Acoustic tuning of the new DePaul University School of Music Concert Hall. Brian Corry (Kirkegaard Assoc., 7733 Forsyth Boulevard, Ste. 1100, St. Louis, MO 63105, bcorry@kirkegaard.com) and Ian Hough (Kirkegaard Assoc., Chicago, IL)

The Holtschneider Performance Center in Chicago packs an enormous program onto a limited urban site. The largest of the four performance halls in this new home for DePaul University's School of Music is the 535-seat Concert Hall. In response to a limited footprint and a broad program of performance types, Kirkegaard devised a simple but innovative acoustical approach to the room. A single massive plaster cloud hovers in the center of the hall, supporting onstage communication and providing first reflections to the audience. The large gap between the cloud and the perimeter walls integrates the extensive volume above the ceiling with the main volume of the hall. Extensive banners above the cloud are intended to "turn off" the upper volume, effectively converting the hall from a concert hall to a

recital hall. More conventional curtains and banners below the cloud allow further fine tuning of the room's acoustics. The Concert Hall is well isolated acoustically from the rest of the 3 storey music building and exterior city noise by 18" thick concrete walls, concrete roof, and a continuous structural acoustic isolation joint.

1:45

1pAA3. Recent acoustic designs in China, challenges, and design approach. Thomas Scelo, Peter Fearnside, and Peter Exton (Performing Arts, Marshall Day Acoust., 1601, 16/F The Hollywood Ctr., 233 Hollywood Rd., Sheung Wan 0000, Hong Kong, tscelo@marshallday.com)

Marshall Day Acoustics has now completed and commissioned sixteen performing arts venues in China and has another sixteen in design or in construction. It seems an opportune time to reflect on the latest challenges, technically or otherwise and how we have successfully approached them. Projects have become bigger and more complex, yet clients' expectations and requirement for certainty in the outcome have increased. The paper will include three recent examples. The 1,500-seat concert hall and 1,000-seat drama theatre of the Jiangsu Grand Theatre both required on-site design to ensure acoustic requirements were met, in time. The 2,000-seat Qingdao Grand Theatre achieved a very low reverberation time suitable for the use of multiple complex sound systems while maintaining the hard finish look desired by the client. This required both careful predictions and laboratory testing. Finally, the 2,040-seat opera house at Shaanxi Grand Theatre that was delivered in less than three years. Atypical materials and simple design principles were implemented to meet the project timeframe. These cases are examples of successful acoustic engineering and what can be achieved when applying first principles and sound science, backed by simulations, testing and experience.

2:05

1pAA4. Recent acoustic designs in China, commissioning results. Thomas Scelo, Peter Fearnside, and Peter Exton (Marshall Day Acoust., 1601, 16/F The Hollywood Ctr., 233 Hollywood Rd., Sheung Wan 0000, Hong Kong, tscelo@marshallday.com)

Marshall Day Acoustics has recently completed three large performing arts centres in China. Some of the challenges and design approaches are presented in an accompanying paper. These include the Jiangsu Grand Theatre with a 1,500-seat concert hall, 2,300-seat opera house and a 1,000-seat drama theatre. Also included is the 2,000-seat Qingdao Grand Theatre and the 2,040-seat opera house at Shaanxi Grand Theatre. The present paper will provide details on the commissioning measurement results for room acoustics. First the principles of the measurement system and room setup will be briefly presented before the actual measurement data and derived acoustic parameter are revealed.

2:25

1pAA5. Scale model test for the acoustical design of Arts Center Incheon concert hall and acoustic measurement after construction. Kee Hyun Kwak, Hyung Suk Jang, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Seoul 133-791, South Korea, bigdook@hanmail.net)

The Arts Center Incheon (ACI) concert hall was designed to have 1,760 seats, the volume of 18,000 m^3 and the reverberation time of 2.1s with seats fully occupied. To achieve the auditory and visual intimacy, the concert hall had a reverse fan shape with a combination of vineyard and balcony. Various scale models and computer simulation were used to determine the acoustical design including the volume of hall, locations of walls and shapes of finishing materials. Three types of sound diffusers were applied according to wall locations. Appropriate protrusion heights were determined based on sound diffusion targets by each part such as the stage and the sides and back of the auditorium. During the execution design stage, a 1:10 scale model of this concert hall was fabricated to measure acoustic characteristics. An auralization experiment was conducted to assess psycho-acoustic responses to the diffusion design. The acoustic measurements after construction matched the design objectives.

2:45

1pAA6. Sound diffusion design process using scale models of a concert hall and acoustic parameters. Hyun In Jo, Hyung Suk Jang, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., Hanyang University, Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

This study proposed a design process for diffusion walls of the Art Center Incheon concert hall by using scale models and verified the sound diffusion performance in comparison with acoustic parameters. Locations of diffusers and main diffusing surfaces were determined in a 1:50 scale model, and the heights of diffusers in the stage were set and diffusion rates were evaluated according to occupancy density in a 1:25 scale model. A 1:10 scale model enabled the real acoustic characteristics of the concert hall to be evaluated. This model was used to measure scattering/diffusion coefficients of diffusers in each part and to design the finishing shapes of walls. Horizontal diffusers with the highest protrusion were installed on the lateral walls near the stage in order to orientate scattered reflections. Impulse responses were used to investigate acoustic parameters like RT, EDT, G, and C80, and the number of peak reflection (Np) was calculated to compare diffusion performance. It turned out that, when the amount of diffusion changed, the value of Np increased but the values of RT, EDT, and G and the corresponding relative standard deviation (RSD) decreased. The ratio of 1:25 or higher scale models were efficient in designing the diffusion surfaces of the concert hall. The effectiveness of scale models was verified as the directions, locations, protrusion heights, and areas of diffusers were determined.

3:05–3:20 Break

3:20–4:20 Panel Discussion

Session 1pAB

Animal Bioacoustics: Fish and Marine Invertebrate Bioacoustics II

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

Xavier Mouy, Cochair
JASCO Applied Sciences, 2305–4464 Markham Street, Victoria, BC V8Z7X8, Canada

Invited Papers

1:00

1pAB1. Underwater ecoacoustics as a monitoring tool in freshwater environments. Camille Desjonquères (Univ. of Wisconsin Milwaukee, 3209 N Maryland Ave., Milwaukee, WI 53212, desjonqu@uwm.edu), Fanny Rybak (Université Paris-Sud, Orsay, France), Toby Gifford (SensiLab, Caulfield, VIC, Australia), Simon Linke (Australian Rivers Inst., Griffith Univ., Nathan, QLD, Australia), and Jérôme Sueur (Museum national d'Histoire naturelle, Paris, France)

Biodiversity in freshwater habitats is decreasing faster than in any other environment, mostly due to human activities. Monitoring these losses can help guide mitigation efforts. In most comparative or focal studies, sampling strategies predominantly rely on collecting animal and vegetal specimens. Although these techniques have produced valuable data, they are invasive, time-consuming, and typically have limited spatial and temporal replication. There is therefore a need for the development of complementary methods. As with other ecosystems and landscapes, freshwater environments host animals producing sounds, either to communicate or as a byproduct of their life activity. Animals and processes can be recorded, remotely, by unattended equipment and provide global information on local diversity and ecosystem health. We review practical examples of progress in experimentally addressing six main challenges that freshwater ecoacoustic monitoring faces: (1) associating each sound to its emitter, (2) estimating intra-specific sound variations, (3) evaluating diurnal variation, (4) modeling sound propagation, (5) deriving links between ecological condition and sounds, and (6) developing a repository for freshwater sounds. Passive acoustics represents a potentially revolutionary development in freshwater ecology, enabling dynamic monitoring of biophysical processes to inform conservation practitioners and managers.

1:20

1pAB2. Glass sponge reef soundscapes. Stephanie K. Archer (Pacific Biological Station, Fisheries and Oceans Canada, Pacific Biological Station, 3190 Hammond Bay, Nanaimo, BC V9T 6N7, Canada, Stephanie.Archer@dfo-mpo.gc.ca), William D. Halliday (Wildlife Conservation Society, Whitehorse, YT, Canada), Amalis Riera (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Xavier Mouy (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Matthew Pine (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada), Anya Dunham (Pacific Biological Station, Fisheries and Oceans Canada, Nanaimo, BC, Canada), and Francis Juanes (Dept. of Biology, Univ. of Victoria, Victoria, BC, Canada)

Many structured biogenic habitats are biodiversity hotspots and thus possess unique soundscapes largely driven by the biophony. To date, the vast majority of research has focused on shallow-water biogenic habitats such as coral or oyster reefs. Glass sponge reefs are a deep-water habitat analogous in many ways to shallow-water coral reefs. These reefs are built by three species of hexactinellid sponges which form complex 3-dimensional habitats that support diverse communities of animals. Many soniferous animals, including rockfish, are strongly associated with patches of live-sponge dominated habitat within the reef foot-print. Consequently, we hypothesized that glass sponge reefs and the communities they support would generate unique soundscapes. Beginning in September of 2016, we deployed a series of underwater acoustic recorders on sponge reefs throughout Canada's Pacific continental shelf. Initial results show that recorders located on sponge reefs were significantly louder in the mid- and high-frequency bands (100–1000 Hz and 1–10 kHz, respectively). Additionally, many fish calls were detected in recordings from within sponge reefs, while few fish calls were observed at similar depths in off-reef habitats. We will discuss our understanding of the link between soundscapes and biodiversity in glass sponge reef habitats and the potential application of ecosystem-level monitoring.

1:40

1pAB3. Spatial mapping of the biophony of the fishes living in seagrass meadows. Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France), Cathy Anna Valentini Poirier, and Pierre Boissery (Agence de l'Eau RMC, Marseille, France)

Passive acoustics is well suited to assess the diversity and/or activity of marine animals, particularly if cryptic or difficult to observe as in seagrass *Posidonia oceanica* meadows. The ability to locate the emitters allows not only to detect the presence of a specific sound or specie but also to estimate source levels, the communicative space with respect to ambient noise and anthropogenic impact, the

density of vocalizing animals, their movements and potentially home ranges, etc. Here, we located fishes in seagrass meadows (Calvi, France). Two particular sounds dominated the acoustic scene of seagrass meadows: the: 300 Hz drums of *Ophidion rochei* and the ubiquitous 800Hz /*kwa*/ sound of unknown origin. The wavelengths of these fish sounds range from 5 m to 2 m. During six 48-hours period, we deployed a squared array of 4 hydrophones (20 x 50 x 50 m). Here, we describe the algorithm for automated fish sound detection and 3D localization of the sources and assess their performances on our real data. Over 1000 fish sounds were detected per night. These detections were used to create localization maps and study the spatial distribution of fish sound production, estimate source levels and the density of the vocalizing fishes within the meadow and their limits.

Contributed Papers

2:00

1pAB4. Searching for the FishOASIS: Using passive acoustics and optical imaging to identify a chorusing species of fish. Camille Pagniello, Jack Butler, Gerald L. D'Spain, Jules Jaffe, Ed Parnell, and Ana Širović (Scripps Inst. of Oceanogr., Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. #0205, La Jolla, CA 92093-0205, cpagniello@ucsd.edu)

Marine protected areas have been established off the California coast to ensure the persistence and resiliency of the marine ecosystems found here. Kelp forest habitats, in particular, support a diverse assemblage of fishes, many of which produce sound. From May to September 2017, a low-frequency (325–545 Hz) chorus was recorded near the kelp forests off La Jolla, California. The chorus begins each day approximately a half-hour before sunset and lasts for about 3–4 hours. During these times, spectral levels around 400 Hz increased by approximately 30 dB over, although there is significant day-to-day variability in received level. To identify the chorusing fish species, a Fish Optical and passive Acoustic Sensor Identification System (FishOASIS) was developed, consisting of a four-channel SoundTrap ST4300 acoustic recorder and four Sony α7s II cameras. Frequency-domain beamforming was used on signals recorded by the four-element, 20-m aperture, tetrahedral-shaped array to estimate the location of the fish chorus, which appears to be fairly fixed over time. The chorus also was used as a source of opportunity to measure transmission loss in order to determine whether kelp forests can act as acoustic refuges by sufficiently attenuating chorusing sounds. [Research supported by California Sea Grant and NSERC Postgraduate Scholarship-Doctoral.]

2:15

1pAB5. Benthic biophonic assemblages, their environmental divers, eco-acoustic scores at the level of the Western Mediterranean basin, and their implications for large-scale ecosystem monitoring. Lucia D. Iorio, Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Cathy Anna Valentini Poirier, and Pierre Boissery (Agence de l'Eau RMC, Marseille, France)

Benthic invertebrate assemblages are known to produce an important biophony composed of short transient sounds emitted while hunting, feeding, moving, for territorial defense, etc. Although they exhibit dial variations, these sounds are present year-round, 24 hours a day and have the potential to provide information on the habitat and organism-environment relationships. Here, we describe benthic invertebrate sounds (BIS) of two key Mediterranean habitats, coralligenous reefs and seagrass meadows. Then, we assess the environmental drivers of BIS variability for each habitat. 129 400 000 BIS sampled in the summers of 2015–2017 from 135 recording sites over more than 1000 km coastline were used for this study. Each sound was characterized by 28 acoustic features. Principal Component Analyses were performed to identify the most contributing features in terms of diversity and intensity. To quantify and characterize invertebrate assemblages at each site, we defined an eco-acoustic score based on BIS abundance, diversity, and intensity. These scored acoustic assemblages were then tested for environmental drivers such as site, biocenosis, depth, ecosystem health, anthropogenic pressures, etc. We hereby provide an exhaustive ocean basin-wide picture of the benthic biophony and discuss its implications as environmental proxies.

2:30–2:45 Break

2:45

1pAB6. Using passive acoustics to localize vocalizing oyster toadfish (*Opsanus tau*). Rosalyn Putland, Alayna Mackiewicz, and Allen F. Mensinger (Dept. of Biology, Univ. of Minnesota Duluth, 1035 Kirby Dr., Duluth, MN 55812, rputland@d.umn.edu)

Identifying where fish inhabit is a fundamentally important topic in ecology and acoustic tools can help management to prioritize acoustically sensitive times and areas. In this study, passive acoustic monitoring is presented as a viable tool for monitoring the positions of vocalizing fish species, like the oyster toadfish. Time of arrival differences (TOADs) of sound recordings on a four-hydrophone array were used to pinpoint the location of male oyster toadfish, *Opsanus tau,* a sedentary fish that produces boatwhistle vocalizations to attract females. Coupling the TOAD method with cross correlation of the different boatwhistles, individual toadfish were mapped during three-hour periods at dawn, midday, dusk, and midnight to examine the relationship between temporal and spatial trends. Seven individual males were identified within 24.2 m of the hydrophone array and up to 18.2 m of the other individuals. The advantages and disadvantages of using the TOAD method to localize individual fish will be discussed. Additionally, preliminary data on how individual toadfish respond to the anthropogenic sound of passing motorized vessels as well as conspecific boatwhistles will be introduced.

3:00

1pAB7. Automatic detection and localization of croaker's fish calls using beamforming. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@mail.tohoku-gakuin.ac.jp), Kazuki Yamato (Gunma Univ., Sendai, Japan), Ryuzo Takahashi, Tomohito Imaizumi (NRIFE, Fisheries Res. and Education Agency, Kamisu-shi, Ibaraki-ken, Japan), and Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan)

Many kinds of fish, including croaker, produce species-specific low-frequency sounds associated with courtship and spawning. A recording system was used to monitor underwater fish call sounds. We have proposed the method to detect croaker calls from data measured by a single hydrophone. However, it was difficult to detect the desired calls at a high detection rate because of a low signal-to-noise ratio. We proposed the method using beamforming to improve the detection rate. At first step, fish calls are detected from data measured on one hydrophone using the previous method, which detects calls automatically using the acoustic features, that is, duration and periodicity. At second step, additional calls are detected by beamforming the four-channel data. At third step, detected calls are localized by using the time differences of arrivals. It was clarified that the detection rate using the proposed beamforming method was higher than that under the previous method with a single data channel. In addition, it was shown that fish calls could be localized from the acoustic data measured during several weeks. Therefore, this method could monitor sounds from croaker in the ocean.

3:15

1pAB8. Complexity-entropy based approach for detection of fish choruses. Shashidhar Siddagangaiah (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taipei 106, Taiwan, shashi.18j@gmail.com)

Increasing anthropogenic noise around the world ocean are affecting the marine ecology. Recently, acoustic indices (AI) were utilized to quantify

the biophony in the marine soundscape. However, these AI's employed in complex marine environment, dominated by several anthropogenic and geophonic sources are yet to be understood. In this study, we have introduced a method based on complexity-entropy (*C-H*) for detection of biophonic sounds originating from fish chorus. The fish chorus detection performance of *C-H* was compared with AI's such as acoustic complexity index (ACI), acoustic diversity index (ADI), and bioacoustics index (BI). We have utilized the data collected at Changhua (A1) and Miaoli (N1). During the Spring of 2016 and 2017, the region N1 was exposed to continual shipping activities, due to which there was ~10 dB increase in the low frequency (5–500 Hz) noise levels. This enabled us to evaluate the fish chorus detection performance of various AI's and *C-H* method, and the robustness in the presence and absence of shipping activities. The results presented in this study shows that, during the fish chorusing hours, the introduced entropy is positively correlated with Pearson's correlation coefficient (P_{cc}) > 0.95 and complexity is anticorrelated with P_{cc} < -0.95. Therefore, the introduced *C-H* method has potential implication in efficient detection of fish chorus and overcome the limitations confronted by AI's such as ACI, ADI, and BI.

3:30

1pAB9. Sounds from the Amazon: Piranha and prey. Rodney A. Rountree (The Fish Listener, 23 Joshua Ln., Waquoit, MA 02536, rrountree@fishecology.org) and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

The underwater soundscape of an upper tributary of the Amazon River was studied in a four-week survey from 3 to 26 July 2012 within the Pacaya-Samiria National Reserve, Peru. Over 550 individuals representing over 70 species of fishes were auditioned for sound production. In addition, over 641 minutes of natural sounds from the river were recorded from 22 sites. We demonstrate that closely related species of piranha can be distinguished by their hand-held disturbance sounds. Similar piranha sounds were recorded in the wild at locations where piranha were known to be actively feeding. Sounds of catfishes and other fishes were significantly more frequent at piranha feeding sites. Thus, piranha sounds appear to be excellent indicators of local piranha feeding activity and suggest that passive acoustic monitoring (PAM) can be an effective tool for studies on piranha behavior, feeding activity, and impact on prey fishes.

3:45

1pAB10. Buzzing sounds used as a mean of intra-specific interaction during agonistic encounters in male European lobsters (*Homarus gammarus*)? Youenn Jézéquel (Laboratoire des Sci. de l'Environnement Marin, UBO, CNRS, IRD, Ifremer, LIA BeBEST, UMR 6539, IUEM, 12 Rte. de penhuel, Plouzané 29280, France, youenn.jezequel@univ-brest.fr), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Jennifer Coston-Guarini (Laboratoire des Sci. de l'Environnement Marin, UBO, CNRS, IRD, Ifremer, LIA BeBEST, UMR 6539, IUEM, Plouzané, France), Jean-Marc Guarini (UPMC (Paris-6), UMR 8222 LECOB, Observatoire Océanologique de Banyuls sur Mer, UPMC, Paris, France), and Laurent Chauvaud (Laboratoire des Sci. de l'Environnement Marin, UBO, CNRS, IRD, Ifremer, LIA BeBEST, UMR 6539, IUEM, Plouzané, France)

Passive acoustics is a useful non-invasive tool to collect behavioral information in marine species. This is the case for temperate crustaceans which are known to emit a large variety of sounds through diverse mechanisms. But despite numerous studies in tanks, little is known about their ecological meaning, particularly for decapods of high commercial interest. When stressed by handling, the European lobster (*Homarus gammarus*) vibrates its carapace and produces low frequency "buzzing sounds" that can be characterized in tanks. In this presentation, we discuss a straightforward experimental approach to investigating the role of these buzzing sounds in male European lobsters by combining passive acoustics and behavioral analysis (video and accelerometry). We recorded sound and video simultaneously during agonistic encounters. Based on the video, an ethogram was created with a total of 30 behaviors regrouped by agonistic levels. During agonistic encounters, European lobsters emitted buzzing sounds in association with stressful events such as claw grasping or tail flipping. Our results suggest that these sounds may be used by *H. gammarus* to maintain dominance around its shelter.

4:00

1pAB11. Describing new fish sounds from the northeast Pacific: A quantitative approach. Amalis Riera (Biology, Univ. of Victoria, School of EOS, Bob Wright Ctr. A405, UVic, Victoria, BC V8P5C2, Canada, ariera@uvic.ca), Rodney A. Rountree (none, Waquoit, MA), and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

In order to identify fish sounds in British Columbia soundscapes, we need catalogues of validated fish sounds from Pacific species. These catalogues will help in comparing validated examples to unknown sounds found in long-term autonomous recordings. In addition, it is important that the sounds be described quantitatively for accurate identification and comparison. Our goal is to contribute to building such catalogues and to fill knowledge gaps in fish acoustic behaviour to support studies on the impact of anthropogenic noise on Pacific fishes. Since distinguishing sound sources in the ocean is extremely difficult, we are collaborating with aquaria, marine science centers and ocean-based aquaculture facilities to monitor, audition and record the acoustic behaviour of captive and semi-captive fish species. To date, we have acquired over 3,000 hours of recordings in these conditions. Sounds that we have already validated include Arctic cod (*Boreogadus saida*), walleye pollock (*Gadus chalcogrammus*), and grunt sculpin (*Rhamphocottus richardsonii*) vocalizations. Here, we will present a description of these newly validated sounds.

4:15

1pAB12. Acoustic recordings of Pacific salmon (*Oncorhynchus* spp.) from a hatchery on Vancouver Island. Kelsie Murchy (Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3N5, Canada, kmurchy@uvic.ca), Xavier Mouy (School of Earth and Oceans Sci., Univ. of Victoria, Victoria, BC, Canada), and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

Sound production in fish has been documented; however, the diversity of species that create sound is not fully understood. Pacific salmon (*Oncorhynchus spp.*) are ecologically, economically, and culturally important in the northeast Pacific. Recent declines in specific species and stocks have increased their public interest. Other species from the family Salmonidae produce sounds, but there is little evidence of any species of Pacific salmon producing sounds. Recording salmon in the wild would be difficult but local hatcheries allow for a unique opportunity to listen for salmon sounds in a semi-natural environment. Chinook salmon (*O. tshawytscha*), pink salmon (*O. gorbuscha*), and coho salmon (*O. kisutch*) were recorded using two stationary acoustic recorders that were deployed at a salmon hatchery in Qualicum beach Vancouver Island, British Columbia, for three consecutive weeks in September and October 2017. Audio files were collected in 5 minute subsections and examined for salmon sounds. Here, we present spectrograms and time-frequency composition of potential sounds found for Chinook, pink, and coho salmon. All sounds were then compared to sounds produced by other soniferous fish and to the hearing range of Pacific salmon.

Session 1pAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Arctic Acoustical Oceanography II

Peter F. Worcester, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225

Hanne Sagen, Cochair
Polar Acoustic and Oceanography Group, Nansen Environmental and Remote Sensing Center, Thormøhlensgt 47, Bergen 5006, Norway

Mohsen Badiey, Cochair
University of Delaware, Newark, DE 19716

Contributed Papers

1:00

1pAO1. Arctic ambient noise statistics at the edge of the Canada Basin. Christopher Whitt, Bruce Martin (JASCO Appl. Sci., 202-32 Troop Ave., Dartmouth, NS B3B1Z1, Canada, christopher.whitt@jasco.com), Sean Pecknold (DRDC Atlantic, Dartmouth, NS, Canada), Mohsen Badiey (Univ. of Delaware, Newark, DE), and Aidan Cole (JASCO Appl. Sci., Halifax, NS, Canada)

Knowledge of ambient noise is important for the design and operation of acoustic observation and communication systems. Several concurrent year-long acoustic datasets from the Canada Basin Acoustic Propagation Experiment (CANAPE) were collected in 2016–2017. Selected data were analyzed to investigate soundscape temporal and spatial characteristics in an area on the western slope of the Chukchi shelf and Canada Basin, north of Barrow Alaska. From October 2016 to October 2017, four 8-element vertical arrays were deployed in water depths between 100 and 300 m. Each array recorded at approximately 15% duty cycle at several sample rates between 4000 Hz and 64000 Hz and at 24-bit resolution. One-minute, 10-second, and 1-second root-mean-square sound pressure levels were computed to generate ambient noise statistics and summarized in empirical probability density functions (PDF) for various bands. Fit functions for ambient noise were determined for each empirical PDF. Impulse detection was used to investigate potential correlation of ice cracking with ambient noise. A multivariate correlation of ambient noise levels was performed with several environmental parameter covariates, including ice cover, wind speed, and air temperature. These correlations may form the basis for predictive models for ambient noise modeling in the arctic.

1:15

1pAO2. Results from the "Arctic ocean under melting ice" acoustic thermometry experiment. Espen Storheim, Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen 5006, Norway, Espen.storheim@nersc.no), Matthew Dzieciuch, Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA), Eva Falck (Geophysical Inst., Univ. of Bergen, Longyearbyen, Norway), and Florian Geyer (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

The Fram Strait is the main gateway for heat exchange between the Arctic Ocean and the Atlantic Ocean, hence estimating the heat transport is important to understand the on-going climate change. An acoustic system for acoustic thermometry was developed under the ACOBAR project. Results from this experiment showed that it was important to monitor the heat content in the north-going West-Spitzbergen current and the south-going East-Greenland current separately. Additionally, the complex oceanographic conditions in this region make it difficult to separate the different arrivals of the acoustic signals in the time domain. The UNDER-ICE experiment, funded by the Research Council of Norway, is the third acoustic thermometry experiment carried out by NERSC in the Fram Strait. Five moorings were deployed from 2014 to 2016, monitoring the north-going and the south-going currents separately. Transmissions were made every third hour for two years along 8 transects, and two moorings were augmented with additional oceanographic sensors. Results from the processing and analysis of the acoustic data are presented, including time series of the depth-range average temperature along the different transects. Oceanographic measurements and a comparison between acoustic observations and modeling results are also presented.

1:30

1pAO3. Using a regional ocean model to understand the structure and sampling variability of acoustic tomography arrivals in Fram Strait. Florian Geyer, Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Nansen Environ. and Remote Sensing Ctr., Thormølens Gate 47, Bergen 5006, Norway, florian.geyer@nersc.no), and Bruce Cornuelle (Scripps Institution of Oceanogr. UCSD, La Jolla, CA)

A regional ocean model for Fram Strait allows to understand the variability and structure of acoustic tomography arrivals. The eddy-permitting model (52 vertical layers and 4.5 km horizontal resolution) was evaluated using long-term moored hydrography data and time series of depth-range averaged temperature obtained from the inversion of acoustic tomography measurements. Geometric ray modelling on the ocean model fields reproduces the measured arrival structure of the acoustic tomography experiment. The combination of ocean and acoustic model gives insights into acoustic propagation during winter and spring. Overlapping arrivals coming from different vertical angles can be resolved and explained. The overlapping arrival of sound channel rays and bottom-reflected rays has implications for the inversion of tomography data in Fram Strait. The increased knowledge about the ray-length variations of bottom-reflected rays is valuable information for choosing appropriate observation kernels for the data assimilation of acoustic tomography data in Fram Strait.

1:45

1pAO4. Effect of sea-ice parameters on acoustic propagation in different Arctic scenarios. Gaute Hope, Hanne Sagen, and Halvor Hobæk (Polar Acoust. and Oceanogr., Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen 5006, Norway, gaute.hope@gmail.com)

Depending on the setup and purpose of an acoustic propagation system (communication, navigation, tomography, seismic exploration) in ice covered areas, different parameters of the sea-ice are more significant in the interaction of acoustic waves with the ice. In homogeneous ice, with a source close to the surface, significant energy can be transferred to flexural waves and other ice-coupled modes. The modes in the ice are strongly dependent on the elastic parameters of the ice, as well as the thickness of the ice-plate. Additionally, discontinuities in the ice are very important to their propagation. In the inhomogeneous and fractured pack ice in the Arctic Ocean, where the cold water layer results in an acoustic surface channel, the majority of the acoustic waves will interact with the sea-ice at high incidence angles. Here, the roughness of the underside is more significant. Measurements from seismic experiments in an ice-covered fjord on Svalbard as well as long-range propagation in the ice-covered Fram Strait, together with simulations using OASES, are shown and used to gauge the effect of the different ice parameters. Although there is insufficient foundation for interpolation between the two extremes, the characteristics of the scenarios which determine the dominant ice-parameter are described.

2:00

1pAO5. A short history of ONR ice camps: 1978–1994. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

The Office of Naval Research sponsored Arctic field programs almost every year from 1978 to 1994 during the height of the Cold War. Almost all of them had an acoustics component coupled with observations for physical oceanography, geoacoustics, plate tectonics, ice mechanics, and used both active and passive methods. In 1978, these started with emphasis on basin reverberation and ended in 1994 with trans Arctic Ocean tomography. They had acronyms from CANBARX (Canadian Basin), FRAM I—IV, MIZEZ (Marginal Ice Zone), PRUDEX (Prudoe Bay) CEAREX and SIMI/TAP (Sea Ice Mechanics/Trans Arctic Propagation). The author was the chief scientist for most of these programs and will provide an overview of ONR's efforts in the Arctic during the Cold War. [Work supported by ONR.]

2:15

1pAO6. Characteristics of the Arctic environment in the southern Beaufort Sea from Ice Exercise data. John E. Joseph, D. Benjamin Reeder, and Derek R. Olson (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jejoseph@nps.edu)

In early March of 2016 and 2018, the Naval Postgraduate School participated in the biennial naval Ice Exercise (ICEX) conducted in the southern Beaufort Sea. Oceanographic and acoustic data sets collected near the ice camps during both events are compared. While the drift track of the ice camp during ICEX-18 was approximately one-degree south of the track in ICEX-16, there are important similarities in the oceanographic structure shown in both datasets. These characteristics have significant impacts on sound propagation in the region and may affect the performance of acoustic systems such as naval sonar and UUV navigation. Of particular interest are properties at the interface between the cold, fresh surface layer and the contrasting warm, saline Pacific Summer Water (PSW) that lays immediately below it. Sensors indicate turbulent mixing of high-spice and low-spice water occurs at this interface. PSW also sets up a stable subsurface sound channel with Pacific Winter Water and Atlantic Water layers below it. Strength of the sound channel varies from year to year; however, historical data from this region indicates an increasing trend. Other oceanographic features found in the upper 200 m of the under-ice water column in our 120-kHz echosounder dataset are discussed.

2:30

1pAO7. Spectral, spatial, and temporal characteristics of underwater ambient noise in the Beaufort Sea in 1994 and 2016. Rui Chen (Mech. Eng., MIT, 77 Massachusetts Ave., Rm. 5-223, Cambridge, MA 02139, ruic@mit.edu), Andrew J. Poulsen (Appl. Physical Sci., Lexington, MA), and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

Climate induced changes in the Arctic Ocean have severely impacted underwater acoustic communication and navigation; understanding underwater noise characteristics is critical to improving the performance of these operations. Ambient noise from the Beaufort Sea recorded in experiments more than 20 years apart (SIMI94 and ICEX16) are compared to determine differences that may be attributed to the region's rapidly changing environment. Both datasets are collected at spring time under packed ice conditions with 32 element vertical line arrays. Spectral comparison shows noise within 20–300 Hz band is 30–40 dB louder in 1994 than 2016, suggesting the ice cover during SIMI94 was more acoustically active. Beamforming results show ambient noise vertical directionality is focused near the horizontal during SIMI94 but more spread in elevation during ICEX16, with a robust noise notch at the horizontal. Numerical modeling demonstrates that this difference may be attributed to ambient noise during ICEX16 being dominated by surface noise sources at discrete ranges rather than the historical assumption of a continuous and uniform distribution of sources. Temporal statistics of transient ice events show more transient activity during SIMI94 and appear to support the new proposed surface source distribution for ICEX16. [Work supported by ONR and DARPA STO.]

2:45

1pAO8. Multipath arrival tracking for marine vehicles utilizing pattern recognition. Jordan Fouquette and Henrik Schmidt (Mech. and Ocean Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm 5-204, Cambridge, MA 02139, henrik@mit.edu)

In recent years, interest in the Arctic Region has been steadily growing as it has become more accessible due to continued ice recession. Extreme temperatures and significant ice cover have created a unique and challenging acoustic environment. At increased distances, individual ray path data become inconsistent due to improper ray path identification and fading. Through the use of pattern recognition, a multipath arrival tracking algorithm was developed to utilize the unique characteristics associated with each individual ray path for long range tracking purposes. This tracking algorithm analyzes the amplitude and arrival time patterns amongst all individual ray paths in order to accurately identify each ray path as scattering and fading occurs, thereby increasing range-tracking capabilities. This becomes especially useful in the Arctic Region as contacts of interest can be tracked regardless of their position above, below, or within the Beaufort Duct—a sound duct from 100 to 200 m depth. Simulations covering the numerous depth combinations of sources and receivers with respect to the Beaufort Duct highlight the effectiveness that is presented by utilizing multipath arrival data. The developed algorithm takes advantage of these unique patterns in order to provide a tracking capability for marine vehicles to employ. [Work supported by ONR.]

3:00

1pAO9. Double-difference tracking of migrating Bowhead Whales using autonomous vector sensors 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), Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, 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 autonomous vector sensors, 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 sensors were not precisely time-synchronized, making relative time-of-arrival information unreliable 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. This same concept has also been used to track fin whales on a seafloor seismic network. Here, the double-difference method is applied to previously localized bowhead whale calls in order to improve their relative positions. 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 may allow tracking of individual whales, which would provide insight into the function of these calls.

3:15–3:30 Break

3:30

1pAO10. Estimating North Pacific Right Whale calling depths in the Bering Sea via modal dispersion. 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), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Margaux Thieury (The École Nationale Supérieure de Techniques Avancées de Bretagne, Brest, France), Aileen Fagan (U.S. Coast Guard Acad., New London, CT), Chris Verlinden (U.S. Coast Guard Acad., San Diego, CA), Catherine L. Berchok, and Jessica Crance (Marine Mammal Lab., NOAA 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 at the 50 m isobath. 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 then used to estimate range, depth, and propagation environment. When plotted as a function of range and frequency, MMP ambiguity surfaces reveal the existence of large sound speed gradients in the sediment, along with strongly downward-refracting sound speed profiles during certain summer/fall seasons. Refractive propagation effects from these profiles can induce dramatic changes in signal structure that required new developments in "warping" techniques. 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. Current work is examining whether consistent differences exist in the calling depths of bowhead, humpback, and right whales, which could potentially provide a feature for species classification of ambiguous calls. [Work sponsored by NPRB.]

3:45

1pAO11. Acoustic response of a migrating bowhead whale population to open-ocean ambient noise levels. 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), Katherine H. Kim, and Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, CA)

Automated and manual acoustic localizations of migrating bowhead whale calls in the Beaufort Sea were used to examine their acoustic response to changes in wind-driven, continuous, ambient noise levels. At low noise levels, the population's source levels and calling rates increased with increasing noise level, with the source level distribution adjusting to maintain a consistent functional detection range, estimated to be between 20 and 60 km. However, once noise levels exceeded the 75thpercentile of their long-term distribution, source level increases failed to keep pace with further increases in noise level, thereby reducing the population's detection range and associated communication space. Call production rates, on the other hand, continued to increase even up to the highest noise levels. Migrating bowhead whales thus attempted to maintain long-range call detectability by adjusting their source level and calling rate. Beyond a certain noise level, whales cannot increase their source levels, but do continue to increase their calling rate. The results provide context for interpreting the effects of industrial noise on bowhead whale acoustic behavior; for example, distant airgun signals stimulate an increase in mean call production rate equivalent to a 26 dB increase in natural ambient noise levels. [Work sponsored by NPRB.]

4:00

1pAO12. Assessment of safety zones for marine mammals at noise exposure from industrial activity in the Russian Arctic. Alexander Vedenev (P.P. Shirshov Inst. of Oceanology of the Russian Acad. of Sci., 36 Nakhimovsky Prospekt, Moscow 117997, Russian Federation, vedenev@ocean.ru), Konstantin Avilov (A.A. Blagonravov Inst. for Machine Sci. of the Russian Acad. of Sci., Moscow, Russian Federation), and Andrey Lunkov (Prokhorov General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation)

E&P of hydrocarbon along with shipping are the main sources of man-made noise in the ocean. But very few information on the noise from industry activities in Arctic is currently available. Over the next decade we expect the increasing level of anthropogenic sound in the Russian Arctic due to planned large scale construction of Oil&Gas projects including the Shtokman project in the Barents Sea, one of the world's largest natural gas deposits. Noise from seismic surveying, pile driving and construction can damage hearing or disquiet marine mammals. To protect marine mammals from pulsed noise of seismic survey, pile driving and chronic continuous noise at different stages of the Shtokman project construction we estimated the sizes of safety zones with SPL thresholds of 180 dB (zone of injury) and 120 dB for continuous noise (behavior disturbance). Modeling of the transmission loss was done by Normal Mode and PDPE models based on the environmental characteristics in the Barents Sea and real spectra of noise sources—construction vessels and seismic airgun array. The results demonstrate changes in the Safety Zones footprints depending on the stage of construction, season, specific environmental conditions like ice cover interface or presence of gas saturated sedimentary layer.

4:15

1pAO13. Monitoring the Arctic acoustic environments with the International Quiet Ocean Experiment. Philippe Blondel (Phys., Univ. of Bath, Claverton Down, Bath, Avon and NE Somerset BA2 7AY, United Kingdom, p.blondel@bath.ac.uk) and Hanne Sagen (NERSC, Bergen, Norway)

The northern high-latitude regions, including the Arctic Ocean, are becoming increasingly important as a result of global warming and their growing economic and political interests. Sea ice reduction is facilitating resource exploration, marine transport, and other economic activities in the regions. Warming waters lead to shifts in marine ecosystems and in soundscapes. Exploitation of resources in the Arctic is expected to grow in the coming decades, offering new opportunities for marine and maritime industries. To measure the environmental impact of ocean noise at a variety of spatial and temporal scales, the International Quiet Ocean Experiment (http://iqoe.org/) established in late 2017 a working group on Arctic Acoustic Environments. The first activities of the Working Group are focusing on identifying locations and times of existing and past acoustic studies in the Arctic Ocean, and synthesise the state-of-the-art on sounds, past, present, and future in the Arctic Ocean. WG activities at the Arctic Observing Summit 2018 (Davos, Switzerland) are linking with indigenous communities and other local stakeholders, to address emerging trends in marine transport and Arctic resource exploitation, and to plan for where/when the optimal acoustic surveys could be, and what metrics they should prioritise.

4:30

1pAO14. Multi-year measurements of the underwater noise field directionality in the shallow Beaufort Sea during open-water and drifting ice flow conditions. 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), Katherine H. Kim, and Alexander Conrad (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 fall (westward) bowhead whale migration. DASARs have one omnidirectional pressure sensor and two orthogonal particle motion sensors, which permit instantaneous measurements of the azimuths of both transient signals and continuous noise between 20 and 500 Hz. The lack of significant shipping or industrial noise in this region provided 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 space across seven seasons. The dominant directionality of the diffuse ambient noise field varied strongly with frequency and was 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. The influence of local drifting floes and sheets of ice on this directionality is also examined. [Work sponsored by ONR.]

MONDAY AFTERNOON, 5 NOVEMBER 2018 SIDNEY (VCC), 1:00 P.M. TO 5:15 P.M.

Session 1pBA

Biomedical Acoustics and Physical Acoustics: Therapeutic Ultrasound Transducers

Tatiana D. Khokhlova, Cochair
Harborview Medical Center, University of Washington, 325 9th Ave., Box 359634, Seattle, WA 98104

Adam D. Maxwell, Cochair
University of Washington, 1013 NE 40th St., Seattle, WA 98105

Invited Papers

1:00

1pBA1. Therapeutic ultrasound transducers conformal to medical needs. Cyril Lafon, Apoutou N'Djin, David Melodelima, Françoise Chavrier, Raphaël Loyet, and Jean-Yves Chapelon (LabTAU, INSERM, 151, cours Albert Thomas, Lyon 69424, France, cyril.lafon@inserm.fr)

The use of ultrasound for therapy has seen a growing interest since the nineties and the development of endorectal transducers for ablating thermally prostatic tissues. Different devices are currently under clinical evaluation for the treatment of many disorders. Numerical modelling plays a significant role in these developments. For each clinical need, it allows defining the geometry of the transducer and the exposure conditions. The technology for manufacturing the transducer is chosen according to the required acoustic output, density of elements, frequency bandwidth, and medical use (endocavitary, disposable, implantable…). The imaging modality for treatment monitoring is also critical and should be considered in the design process. The goal of this presentation will be to list the various challenges faced during the design of a therapeutic transducer, discuss some solutions that were proposed, and illustrate the process with different examples.

1:20

1pBA2. Rapid prototyping for therapeutic ultrasound transducers. Timothy L. Hall (Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109, hallt@umich.edu) and Adam D. Maxwell (Univ. of Washington, Seattle, WA)

Introduction: Piezoelectric materials are readily available from many suppliers at low cost. Finished ultrasound transducers, however, are significantly more expensive particularly for custom designs. Acoustics research often requires many iterations before a successful experiment design is achieved. This talk will present examples and what we've learned from ten years experience at the University of Michigan constructing transducers with rapid prototyping. Methods: Selective Laser Sintering, Stereolithography, and Fused Deposition Modeling techniques were all tested for transducer construction. Acoustic properties (sound speed, absorption, and impedance) were measured for various materials in each category. Ageing and degradation due to water ingress were assessed as well. Results: Housings for electrical/water isolation were successfully constructed using each technology. Professional quality machines yielded sufficient precision to create focused array structures up to 5 MHz. A majority of stereolithography materials showed very poor water compatibility degrading with exposure over time. Most materials have too low acoustic impedance for good matching layers. Particle filled stereolithography resins have increased density and sound speed that raise the impedance to more suitable values. Acoustic lenses were successfully constructed from stereolithography materials allowing focused transducers to be constructed from low cost flat piezoelectric materials.

1:40

1pBA3. Dual-mode capacitive micromachined ultrasonic transducer arrays for high intensity focused ultrasound and imaging. Ji Hoon Jang, Chienliu Chang, Morten F. Rasumssen, Kevin Brenner, Quintin Stedman (Stanford Univ., Edward L. Ginzton Lab Box 131, Stanford, CA 94305, qstedman@stanford.edu), Arif S. Ergun (TOBB Univ. of Economics and Technol., Stanford, CA), and Butrus T. Khuri-Yakub (Stanford Univ., Stanford, CA)

High Intensity Focused Ultrasound (HIFU) therapy requires imaging guidance to ensure that the correct area is treated. Guidance may be provided by magnetic resonance imaging or ultrasound imaging. With piezoelectric transducers, ultrasound imaging guidance is typically done with a separate transducer array since it is difficult to obtain sufficient imaging and HIFU performance with the same transducers. However, Capacitive Micromachined Ultrasonic Transducers (CMUTs) can have sufficient bandwidth and efficiency to allow both HIFU therapy and imaging to be done with the same transducer array. Additionally, CMUTs have much lower self-heating than piezoelectric transducers and require less cooling when used for HIFU. This means that CMUT-based HIFU systems can be made smaller than other HIFU systems to date, allowing access to treatment areas that were not previously possible. In addition, using the same array for imaging and treatment eliminates challenges with co-registration. We have constructed a dual-mode HIFU and imaging probe using a 32x32 CMUT array integrated with an application-specific integrated circuit. Ultrasound imaging guided ablation of ex-vivo tissue has been demonstrated with the probe, with rapid switching between HIFU and imaging mode.

Contributed Papers

2:00

1pBA4. Various approaches for designing phased arrays for high-intensity focused ultrasound therapies: From sparse to fully-populated configurations. Oleg Sapozhnikov, Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105; Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Pavel Rosnitskiy (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Leonid Gavrilov (N.N. Andreyev Acoust. Inst., Moscow, Russian Federation)

High-intensity focused ultrasound (HIFU) therapies are often performed using multi-element phased arrays. Independent variation of the amplitudes and phases at the array elements allows electronic steering of the focus and compensation for aberrations. To suppress the formation of grating lobes, the arrangement of the array elements should be non-periodic. Three approaches that have been recently employed in our studies to solve this problem are presented and discussed: a random element arrangement [L.R. Gavrilov and J.W. Hand, IEEE Trans. UFFC 2000], a multi-armed spiral arrangement [V.A. Khokhlova *et al*., Physics Procedia 2016], and recently proposed array design with tight packing of the elements based on the capacity-constrained tessellation [P.B. Rosnitskiy *et al*., IEEE UFFC 2018]. The efficiency of two arrays with the same geometric and physical parameters is compared: a 256-element array with a compact 16-spirals layout of circular elements and a fully populated array comprising polygonal elements of equal area. It is shown that for the same intensity at the elements, the fully-populated array provides twofold higher total power while maintaining the same electronic focusing capabilities as compared to the spiral one which can be beneficial for high-power applications such as histotripsy. [The work was supported by NIH R01EB7643 and R01EB025187, RFBR 17-02-00261, and Ph.D. student stipends from the "Basis" Foundation and of the President of Russia SP-2644.2018.4.]

2:15

1pBA5. High throughput system for preparing samples for genomic and epigenetic assays. Thomas Matula, Karol Bomsztyk, Brian MacConaghy, and Adam D. Maxwell (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu)

Genomic and epigenomic therapies hold great promise for "fixing" faulty genes. In genomics, DNA is sequenced to determine the order of the nucleotides, while in epigenetics, immunoprecipitation of chromatin is used to locate specific proteins of interest. The first step for these therapies is to identify the specific sequence or gene responsible for the disease. To prepare samples for these assays, DNA or chromatin is sheared into fragments, most often using cavitation. However, standard tools such as ultrasonic horns can be highly inconsistent, leading to poor assays. In order to overcome the problems of inconsistent fragmentation, we designed and built a transducer array (free field pressures $P^+ = 30$ MPa, $P^- = 12$ MPa) capable of processing samples directly in 96-well microplates. An array of transducers is mounted below a microplate, coupled to a lens array that focuses acoustic energy into each well of a microplate. Intense cavitation is generated in each well. After treatment, gel electrophoresis shows that DNA and chromatin are fragmented to a range of sizes extending as low as 100 base pairs (34 nm). Importantly, this process results in consistent fragmentation that leads to consistent repeatable assays. [Funded by NIH R33CA191135, R21GM111439, R01DK103849 and Life Sciences Discovery Fund #12330479.]

Invited Papers

2:30

1pBA6. Low frequency (20 kHz), patch-like ultrasound applicator for chronic wound treatment. Peter A. Lewin, Olivia Ngo, Evan Niemann, Vivinya Gunasekaran, Prabagar Sankar, Alec Lafontant, Sumati Nadkarni (School of Biomedical Eng., Sci., and Health Systems, Drexel Univ., 3141 Chestnut St., Bossone 701, Philadelphia, PA 19104, lewinpa@drexel.edu), Rose Ann DiMaria-Ghalili (College of Nursing and Health Professions, Drexel Univ., Philadelphia, PA), Michael Neidrauer, Leonid Zubkov (School of Biomedical Eng., Sci., and Health Systems, Drexel Univ., Philadelphia, PA), Michael Weingarten (Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), and David Margolis (Dermatology, Perelman School of Medicine Univ. of Pennsylvania, Philadelphia, PA)

Chronic wounds, such as venous and diabetic ulcers, cost the U.S healthcare system alone, close to $25 billion annually. Hence, a reduction of healing time directly translates into savings of treatment related expenses. This work describes the implementation of patch-like, un-tethered and clinically viable therapeutic ultrasound applicator. The device uses well-defined non-cavitational and non-thermal levels of ultrasound energy; its peak acoustic output pressure amplitude was intentionally limited to 55 kPa, corresponding to a spatial peak temporal peak intensity of 100 mW/cm^2. A small (n = 8) pilot study targeting diabetic ulcers treatment was performed and

indicated that with its light weight (<20g), and circular (40 mm dia) disk shape architecture this applicator is suitable to be embedded in wound dressing. The average time to wound closure was 4.7 weeks for subjects treated with the active device, compared to 12 weeks for subjects treated with a sham applicator, suggesting that patients with diabetic ulcers may benefit from the proposed treatment. [Work supported by the NINR grant5R01NR015995. The contents of this presentation are solely the responsibility of the authors and do not necessarily represent the official views of the NIH.]

2:50

1pBA7. Treatment monitoring for sonothrombolysis in deep vein thrombosis: Receiver array design. Christopher Acconcia, Ryan M. Jones, and Kullervo Hynynen (Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N3M5, Canada, chrisacc@sri.utoronto.ca)

The current treatment standard for deep vein thrombosis (DVT) is anticoagulation therapy, which does little to address long term morbidity compared to clot removal approaches. High intensity ultrasound can resolve clots by generating bubble clouds that erode thrombi. However, lack of appropriate treatment monitoring is a limiting factor in its widespread adoption. Passive cavitation imaging is capable of high frame rate, volumetric imaging, the combination of which has been shown to be important for monitoring the onset and development of bubble clouds inducing clot erosion (Acconcia *et al.* 2017). The results from our previous work motivated development of a device tailored to DVT with a relevant geometry (e.g., semi-cylindrical). Transmit simulations for such a device were conducted in a human thigh model (Smirnov and Hynynen 2017) with a design based on the modular transducer technology developed in our lab (Ellens *et al.* 2015). Here, we examine the integration of a sparse, randomly distributed receiver array within a semi-cylindrical arrangement of transmit modules for acoustic-based monitoring. Using a multi-layered propagation model, cavitation sources were localized to the femoral vessel, the accuracy of which depended on the inclusion of phase corrections. The receiver size was shown to be an important consideration with implicit trade-offs between directivity and channel SNR. Volumetric rates of ~1 MHz should be achievable with a modest number of receivers (128) in the presence of experimentally derived noise conditions.

3:10–3:25 Break

Contributed Papers

3:25

1pBA8. Creating uniform ultrasound fields in the brain without an array. Luke A. Richards, Robin Cleveland, and Eleanor P. Stride (Dept. of Eng. Sci., Oxford Univ., Inst. of Biomedical Eng., Old Rd. Campus, Oxford, Oxfordshire OX3 7DQ, United Kingdom, luke.richards@eng.ox.ac.uk)

There has been a recent interest in using patient specific ultrasound lenses to correct for the phase aberration caused by the skull in focused transcranial ultrasound. Here, we apply acoustic lenses to the problem of creating a highly uniform field inside the skull, motivated by the requirements of targeted and ultrasound responsive nanodroplets being developed for the treatment of brain metastases. The phase and amplitude required for uniform sonication was determined from simulations of ultrasound propagation utilising CT data as a basis for the geometry of the patient's skull. From these simulations, silicone lenses were produced, capable of introducing phase inhomogeneity in the incident ultrasound beam to compensate for the effects of transmission through the skull. Experimental measurements with an *ex vivo* skull showed that phase uniformity could be at least partially restored within the skull cavity. Methods for modulating the amplitude of incident ultrasound have also been investigated. Stereolithography was used to directly print in highly attenuating (15 dB/cm at 1 MHz) materials, allowing rapid production of customised and selective acoustic absorbers. A 3D printed transducer backing was also studied, able to suppress transducer output in certain regions, and provide up to 15 dB dynamic range across the surface of a planar ultrasound transducer.

3:40

1pBA9. Enhanced shock scattering histotripsy with pseudo-monopolar ultrasound pulses. Yige Li, Timothy L. Hall, Zhen Xu, and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, 2135 Carl A. Gerstacker Bldg., 2200 Bonisteel Boulevard, Ann Arbor, MI 48109, yigeli@umich.edu)

Shock scattering histotripsy (SSH) involves a complex interaction between positive and negative phases of an acoustic burst to initiate a robust cavitation bubble cloud. To more precisely study these effects and optimize SSH therapy, we constructed a frequency compounding transducer to generate pseudo-monopolar ultrasound pulses. The transducer consisted of 113 individual piezoelectric elements with various resonant frequencies (250 kHz, 500 kHz, 750 kHz, 1 MHz, 1.5 MHz, 2 MHz, and 3 MHz). For each resonant frequency, a nearly 1.5-cycle pulse could be generated. Pseudo-monopolar peak positive pulses were generated by aligning the principal peak positive pressures of individual frequency components temporally so that they added constructively, and destructive interference occurred outside the peak-positive-overlapped temporal window. After inverting the polarity of the excitation signals, pseudo-monopolar peak negative pulses were generated similarly by aligning principal peak negative pressures. Decoupling the positive and negative acoustic phases could have significant advantages for therapeutic applications enhancing precision, avoiding cavitation at tissue interfaces, and reducing the acoustic aperture required for effective therapy. For example, we show that 16 SSH bubble clouds can be generated using only peak positive pulses following a single peak negative pulse. This results in a precise elongated lesion within red-blood-cell phantoms.

3:55

1pBA10. Development of a freely available simulator with graphical interface for modeling nonlinear focused ultrasound fields with shocks. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle; Phys. Faculty, Moscow State Univ., CIMU, APL, University of Washington, Seattle, Washington 98105, verak2@uw.edu), Petr V. Yuldashev, Ilya Mezdrokhin, Pavel Rosnitskiy, Maria M. Karzova (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA; Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Measurement-based modeling is gaining acceptance as a standard tool for characterizing the nonlinear fields of existing therapeutic ultrasound devices and designing new ones. Here, a freely available simulation tool is presented for modeling axially symmetric, strongly nonlinear HIFU beams with shocks in a layered propagation medium such as water and different types of tissue. Two nonlinear wave equations are included in the simulator: the KZK equation generalized to include an equivalent source boundary condition for strongly focused beams and the Westervelt equation in a

nonlinear wide-angle parabolic representation. Both equations are solved in the frequency domain and permit definition of the HIFU transducer as an annular array. The geometrical parameters and power output of the array, electronic focus steering along the beam axis, and acoustic properties of the layered propagation medium can be configured via graphical interface. Visualization and output of various acoustic field parameters such as peak positive and negative pressures, shock amplitude, intensity, heat deposition rate, and total power of a beam are also provided. The simulator can be used for transducers without ideal symmetry through definition of an equivalent radially symmetric source. Widespread availability of such simulation tools will help advance standardized utilization of measurement-based modeling and facilitate the adoption of such approaches for HIFU treatment planning. [Work supported by NIH R01EB7643, R01EB025187, and RSF №14-12-00974.]

4:10

1pBA11. Nonlinear ultrasound fields generated by a 256-element spiral array for boiling histotripsy. Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98104, tdk7@uw.edu), Wayne Kreider, Christopher Hunter, Mohamed A. Ghanem, Bryan Cunitz (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ.,Moscow, Russian Federation), and Vera A. Khokhlova (Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Seattle, WA)

Boiling histotripsy (BH) is a HIFU approach that uses millisecond-long ultrasound bursts with high-amplitude shocks to mechanically fractionate tissue. In this work, the performance of a BH pre-clinical system operating at 1.5 MHz was characterized by hydrophone measurements in water and lesion formation capabilities were tested in *ex vivo* tissue. The system comprises a 256 element array driven by Verasonics Ultrasound Engine with a 1.2 kW external power source enhanced by an additional capacitor bank. The maximum pulse average acoustic power of the system was 2.2 kW (as measured by radiation force balance), sustained for 10 ms with <15% drop in amplitude. At the focus, a fully developed shock of 100 MPa amplitude formed at 275 W acoustic power. The steering range defined at -3 dB in both linear and nonlinear beam was 19 mm laterally and 38 mm axially forming the region where 100 MPa shocks can be achieved with less than twofold increase in acoustic power. Within the axial steering range, shock formation characteristics varied substantially due to changes in the effective transducer F-number. The system was successfully used to produce volumetric BH lesions in *ex vivo* tissue using electronic steering only. [Work supported by NIH R01EB7643, R01GM122859, and R01EB025187.]

Invited Paper

4:25

1pBA12. New international standards for high pressure therapeutic ultrasound fields. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

High pressure levels of strongly focused therapeutic transducers create new challenges for formulating international standards for measurement and simulation. Earlier efforts of Working Group 6, High Intensity Therapeutic Ultrasound and Focusing Transducers, Technical Committee 87 of the International Electrotechnical Commission, resulted in a standard for measuring high power (up to 500 W) and a specification for measuring ultrasound fields at lower amplitudes using existing linear methods and scaling up to higher therapeutic levels. Presently, work is underway on a new document to describe direct approaches for characterizing ultrasound pressure fields in water up to 100 MPa through the use of state of the art optical and robust hydrophones. Because the real goal of our efforts is to determine therapeutic pressure levels in tissue at a target site, we have initiated work on a second hybrid approach which combines characterization of the radiated field close to the transducer in water with nonlinear field simulation programs. In this way the characterization data which preserve the uniqueness of a particular transducer are used as input to a simulator program for realistic field prediction. Our group is addressing how to simulate nonlinear propagation in heterogeneous absorbing tissues accurately and to benchmark simulation software.

Contributed Papers

4:45

1pBA13. Needle hydrophone transfer function model for characterizing therapeutic transducers. Keith A. Wear and Yunbo Liu (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)

A rigid piston model has been developed to predict needle hydrophone transfer functions. The model predicts that "effective" sensitive element diameter depends on the frequency and can exceed geometrical sensitive element diameter by a factor of up to 2.25. The model predicts harmonic distortion for hydrophone measurements of focused pressure waves, which were modeled using nonlinear plane wave theory for spectral characteristics and Gaussian radial pressure distributions. The pressure wave model was validated by comparing to hydrophone (sensitive element diameter: 85 microns) measurements performed in focal planes of source transducers (3.5 MHz, 64 mm diameter, 89 mm focus and 5 MHz, 19 mm diameter, 38 mm focus). Root mean squared difference (RMSD) between the model fit and measured relative strengths of the first four harmonics was 9% (3.5 MHz transducer) and 11% (5 MHz transducer) where Blackstock's sigma parameter (*JASA*, 39, 1019–1026, 1966), which measures extent of nonlinearity, was optimized to fit the data. RMSD between model and measured half width half maxima was 10 ± 5%. Tone bursts from each transducer were measured using needle hydrophones (sensitive element diameters: 200, 400, 600, and 1000 microns). RMSD between model and measured harmonic distortion was 12 ± 4% for the first four harmonics.

5:00

1pBA14. Holographic characterization of therapeutic array transducers with assessment of measurement limitations and robustness. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Seattle, WA), Mohamed A. Ghanem (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Seattle, WA)

Acoustic holography measurements and corresponding numerical projection methods have recently gained acceptance for characterizing the fields generated by high intensity therapeutic ultrasound (HITU) applications. To facilitate the standardization of such measurement-based simulation methods, it is important to understand how practical measurement challenges and limitations impact the results. Toward this end, holography measurements were acquired and analyzed for characterization of two therapeutic array transducers, each with 256 elements. For different focusing configurations, measurements comprised 2D holography scans as well as 1D scans used to independently quantify the focal lobe. The fields projected from holograms were compared to independent 1D scans to evaluate the impacts of hydrophone directivity, non-orthogonality of the axes of the positioner used to move the hydrophone, and overall mechanical stability of measurements. Results demonstrate that directivity effects exceeded those expected based on hydrophone size and altered focal pressures on the order of 10%. Non-orthogonality of the positioner shifted apparent focal locations by measurable distances, which could be confused with small shifts in device fixturing. However, holographic reconstruction of the ultrasound field structure near the focal lobe was robust with respect to directivity and non-orthogonality, which enables compensation by relative calibration of source output levels. [Funding support by NIH R01-EB025187, R01-EB007643, and P01-DK043881.]

MONDAY AFTERNOON, 5 NOVEMBER 2018 RATTENBURY A/B (FE), 1:00 P.M. TO 2:15 P.M.

Session 1pEAa

Engineering Acoustics: Transducer Characterization

Caleb F. Sieck, Chair

Code 7160, NRC Postdoctoral Research Associateship Program, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Contributed Papers

1:00

1pEAa1. Using and selecting small condenser format microphones for impulsive noise measurement. James Weir (Bruel & Kjaer North America, 3079 Premire Pkwy, Ste. 120, Duluth, U.S. 30097, GA, jim.weir@bksv.com)

The small format condenser microphone is the most often used transducer for impulsive measurements of firearms and suppressors. In this paper, the properties and characteristics of these microphones will be detailed and how the selection and use of these condenser microphones will affect the consistency and accuracy of measurements. The goal is to raise awareness and measurement consistency for developers, regulators, and user/hobbyists. The short wavelengths of sound at high frequencies are physically manipulated by the physical design of the transducers. A thorough testing of these products reveal these characteristics and their effect on the accuracy of the resulting measurement. The directivity, frequency response, impulse response, and damping characteristics will be compared. Results will be represented, along with anecdotal stories of real life implementation.

1:15

1pEAa2. Acoustic beam pattern for a push-pull 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 push-pull laser interferometric sound sensor uses two parallel laser beams where both beams are exposed to the sound filled fluid. In arrays of conventional piezoelectric hydrophones, each array element produces a voltage and the beam pattern is obtained by deriving the response as a function of acoustic wave angle in terms of electrical power divided by the maximum response power along the acoustic axis. In a push-pull laser interferometric sound sensor the response function is an optical path difference or phase difference between the two laser beams as a function of acoustic wave angle. In an analogous way, we derive a response function in terms of the optical path difference between two optical line sensors as a function of acoustic wave angle and divide by the response along the acoustic axis. This normalized response function is squared to obtain the beam pattern for a push-pull laser interferometric sensor. We compare this beam pattern to that of a conventional line array.

1:30

1pEAa3. MEMS microphone based on the membrane with bias voltage and FET (field effect transistor) mechano-electrical transduction. Junsoo Kim, Hoontaek Lee, Chayeong Kim, Donghwan Seo (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), San 31 Hyoja-dong, Nam-gu, Pohang, Gyeongsangbuk-do 37673, 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)

Most commercially available 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 the membrane and backplate structure. Proposed 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 electric field. 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). In this study, design and fabrication, performance test of the proposed FET based MEMS microphone are conducted. [Work supported by CMTC, UM15304RD3.]

1:45

1pEAa4. Comparison of acoustic quality according to the types of flat panel display: Direct drive with exciter speaker. Hyung Woo Park (IT, Soongsil Univ., 1212 Hyungham Eng. Building 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee, Kwanho Park (LGDisplay, Paju, South Korea), and Myungjin bae (IT, Soongsil Univ., Seoul, South Korea)

With the technological improvements in the display industry, display panel manufacturers have changed the picture quality from simple high definition to augmented reality or virtual reality. In the previous research, the sound field implementation and sound quality improvement using organic light-emitting diode (OLED) flat panel was introduced. Recently, we have developed a technology for implementing sound in a drive even on a high-quality liquid crystal display. The LED panel has a multi-layer structure, and the backlight unit has a disadvantage of the wavelength transmission structure. In this study, we proposed a method to analyze LCD more effectively through previous studies. Also, we can play left channel and right channel like analysis 2 channel flat panel speakers with one panel. This flat panel speaker consists of direct drive actuator and diaphragm and is used as the outer glass layer of OLED and multilayer LCD. In the LCD panel, we configured the enclosure and adjusted the exciter for the output powers of the exciter were controlled to ensure. Furthermore, the vibration was transmitted to the last flat sheet. We also implemented a clear-sounding direct-drive exciter speaker flat panel display.

2:00

1pEAa5. A study on the optimal speaker position for improving sound quality of flat panel display. SungTae Lee (Res., LGDisplay, 37-8, LCD-ro 8beon-gil, Wollong-myeon, Paju 10844, South Korea, owenlee@lgdisplay.com), Hyung Woo Park (IT, Soongsil Univ., Seoul, Seoul, South Korea), Kwanho Park (Res., LGDisplay, Paju, South Korea), and myungjin bae (IT, Soongsil Univ., Seoul, South Korea)

As a technological improvement in display industry, flat panel manufacturers have changed the screen quality from simple high definition to 5 sensation satisfaction. The sound quality is improved, which is the easiest way to enhance the senses satisfaction. The technology, which in a previous meeting in Minneapolis, is using an exciter speaker. That is, the vibrate panel itself is used to make and play the sound for multimedia information. In this study, the optimum position of the exciter was studied to improve the sound quality of the proposed multi–channel flat–panel speaker. We studied the pattern of the acoustic energy emitted by the exciter and the position and the radiation pattern of the speaker to prevent distortion of the sound when watching the display from a distance. The panel encloses the exciter and the back plane. We also studied the method of construction.

MONDAY AFTERNOON, 5 NOVEMBER 2018 RATTENBURY A/B (FE), 3:00 P.M. TO 4:30 P.M.

Session 1pEAb

Engineering Acoustics: General Topics in Engineering Acoustics

Caleb F. Sieck, Chair
Code 7160, NRC Postdoctoral Research Associateship Program, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Contributed Papers

3:00

1pEAb1. Production method for simulcast in 22.2 multichannel sound broadcasting. Takehiro Sugimoto and Tomoyasu Komori (Sci. & Technol. Res. Labs., NHK, 1-10-11 Kinuta, Setagaya-ku, Tokyo 1578510, Japan, sugimoto.t-fg@nhk.or.jp)

The 8K with 22.2 multichannel (22.2 ch) sound broadcasting is planned to be launched since December 2018 in Japan. NHK is going to simulcast 2-channel stereo and 5.1 surround together with the 22.2 ch sound for backward compatibility with existing home reproduction equipment, e.g., AV receiver and soundbar. However, individual production for each audio format is impractical because it needs many engineers and a lot of production equipment compared with a production in single format. Hence, an efficient production method for simulcast which does not demand additional production resources is required. The first proposal is a tone compensation for downmixing from the 22.2 ch sound to the other audio formats. The energy spectrum of the downmixed signal is compensated to harmonize with that of the 22.2 ch sound. The second proposal is a loudness chase to have the loudness level of the downmixed signal follow that of the 22.2 ch sound. The loudness level of the downmixed signal usually differs from that of the 22.2 ch sound, so that the loudness chase successively adjusts the loudness level without unnatural level change and matches them by the end of the program.

3:15

1pEAb2. Distributed acoustic sensing for near-surface seismic applications. Richard D. Costley, Gustavo Galan-Comas (U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Kent K. Hathaway (U.S. Army Engineer Res. & Development Ctr., Kitty Hawk, NC), Stephen A. Ketcham (U.S. Army Engineer Res. & Development Ctr., Alexandra, VA), and Clay K. Kirkendall (Naval Res. Lab., Washington, DC)

Distributed acoustic sensing (DAS) has been used in the oil and gas industries for vertical seismic profiling (VSP) for approximately the past ten years. Its use for near-surface seismic surveys has also been investigated. This study relates to these types of applications. DAS consists of a buried optical fiber cable with one of its fibers connected to electro-optical instrumentation, i.e., an optical interrogator. The interrogator injects pulses of

coherent light into the fiber and receives backscattered light. Seismic disturbances deform the cable, strain the interrogated fiber, and change the fiber's optical path length. The resulting phase changes of the backscattered light are recorded by the interrogator. These phase change measurements, proportional to strain, are probed through time at spatially sequential (i.e., "distributed") segments along the length of the fiber. The length of each segment is referred to as the gauge length. A single fiber is thus a nominal array of sensors with the gauge length defining the spatial separation. Here, we present DAS signals from ground vibrations caused by impact-hammer and electro-magnetic-shaker sources. The signals were recorded simultaneously by geophones for comparison. Interrogator modifications allowed selectable gauge lengths, 10, 4 and 2 meters. The experimental results demonstrate the benefits of higher resolution.

3:30

1pEAb3. Imaging aeroacoustic sources in a wind-tunnel with a massive array of MEMS microphones. Yinshi Zhou, Vincent Valeau (Institut PPRIME UPR 3346, CNRS-Université de Poitiers-ENSMA, 6 rue Marcel Doré TSA 41105, Poitiers Cedex 9 86073, France, vincent.valeau@univ-poitiers.fr), Jacques Marchal (Institut Jean Le Rond d'Alembert, Sorbonne Université, CNRS, SAINT-CYR-L'ECOLE, France), Régis Marchiano, and françois ollivier (Institut Jean Le Rond d'Alembert, Sorbonne Université, CNRS, Paris, France)

This presentation deals with the identification of aeroacoustic sources in the open section of an anechoic wind-tunnel by using a three-dimensional (3D) array of 256 microphones. The antenna is made of three perpendicular planar arrays enclosing the test-section, and the microphones are digital MEMS microphones. The data processing is based on the beamforming technique associated with a deconvolution method (CLEAN) developed in 3D to improve the spatial resolution. Some preliminary tests with a well-controlled artificial source validate the set-up and the calibration procedure. Some measurements are carried out with aeroacoustic sources generated by the interaction of obstacles with the wind-tunnel flow. The academic case of a cylinder emitting aeolian tones is first considered. A more complex case is then considered with a symmetric wall-mounted NACA airfoil. The different sources (trailing edge noise, tip noise, and junction noise) are successfully identified in a 3D volume in terms of position. The study demonstrates that aeroacoustic source identification is achievable in three dimensions by using a massive array of very cheap MEMS microphones when associated to appropriate data processing techniques.

3:45

1pEAb4. Active control of a highly directional primary source by compact novel secondary sources. Qi Hu and Shiu-Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, KLN, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

The active noise control in free space for a global attenuation is always very challenging, especially for extended and complex radiators. This work focuses on a finite simple line source possessing strong directivity of multi-lobe radiation pattern as the primary source, which is frequently encountered in traffic noise control problems modelling a busy road or the train noise. Conventional secondary sources of compact size radiate the sound wave omnidirectionally, resulting in tangible pressure level increase in off-axis areas. This paper introduces a novel constructed control source, which has reasonably directional sound radiation capability even with a very compact size in low-frequency. Comparing to common directive sources, such as axially oscillating baffled pistons, the novel sources can significantly improve the active control result in a global way.

4:00

1pEAb5. Prediction of noise from an automotive turbocharger centrifugal compressor using three-dimensional computational fluid dynamics. 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)

The noise generated by centrifugal compressors for automotive turbochargers consists of narrow and broadband components. Narrowband noise occurs near integer multiples of the shaft rotational frequency, and the prevailing order is typically equal to the number of main impeller blades (impeller blade-pass frequency). However, for a centrifugal compressor without a ported shroud recirculating casing treatment, such as the current contemporary configuration, broadband noise is dominant, typically in the 4–12 kHz range. This study applies a three-dimensional computational fluid dynamics model to this compressor installed on a steady-flow turbocharger stand. As the compressor flow rate is reduced at constant rotational speed, broadband noise is elevated as a result of deterioration of the flow-field near the inducer (inlet) of the impeller, due primarily to flow separation from the suction surface of impeller blades. Detached eddy simulations provide reasonable agreement with the corresponding experimental measurements in terms of capturing performance, as well as acoustics, thereby shedding insight into this noise source mechanism associated with such flow separation.

4:15

1pEAb6. A study on sudden unintended acceleration estimation engine sounds by vehicle type. Uk-Jin Song (Dept. Information and TeleCommun. Soongsil Univ., Sangdo 1-dong, Dongjak-gu, Seoul 156743, South Korea, imduj@ssu.ac.kr), Seonggeon Bae (Kangnam Univ., Youngin, South Korea), and Myungjin Bae (Soongsil Univ., Seoul, South Korea)

A car is one of the road transport that carries the passenger or freight by transmitting power generated by the engine to the wheel. The automobile has been continuously developed up to modern, so that the power can be transmitted to the wheel with higher efficiency. Recently, however, an accident involving the suspicion of mechanical defect has occurred through the electronicization of internal parts of automobiles. One of them is estimated to be a sudden unintended acceleration. A sudden unintended acceleration is a phenomenon in which the rpm of the engine rises rapidly regardless of the intention of the driver, resulting in a rapid acceleration of the vehicle speed. Accidents involving automobiles are decreasing due to various efforts to reduce the incidence rates, but accidents with the sudden unintended acceleration are on the rise. Therefore, in this paper, we want to analyze the engine sounds of accidents claiming sudden unintended acceleration obtained by a black box by the vehicle type.

Session 1pNS

Noise, Physical Acoustics, Signal Processing in Acoustics, and ASA Committee on Standards: Supersonic Jet Aeroacoustics II

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. An investigation of the interactions between impulsively started and steady supersonic jets. Karthikeyan Natarajan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., EAD, PB 1779, Old Airport Rd., Bangalore 560017, India, nkarthikeyan@nal.res.in), Suriyanarayanan P, Lakshmi Venkatakrishnan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., Bangalore, Karnataka, India), and Sankaran Sathiyavageeswaran (Satish Dhawan Space Ctr., Indian Space Res. Organization, India, Sriharikota, Andhra Pradesh, India)

The ignition of the solid rocket booster strap-on and the following IOP wave have a severe influence on the dynamic as well as the acoustic loads experienced by a launch vehicle during lift-off. The jet from the strap-on interacts with the core jet and its established flow over the jet blast deflector, influencing the acoustic field experienced by the launch vehicle. The existing literature provides very little understanding of such interactions. This work investigates the interaction between the jet from SRB and the core, by simulating them with an impulsively started supersonic jet in close proximity to another established jet. While the flow for the core nozzle was allowed directly from the settling chamber, a novel approach using a quick opening valve was used for initiating the flow through the strap-on nozzle. The evolution of a strong shock wave emanating from the strap-on nozzle and the ensuing vortex ring, their interaction with the core jet flow, etc., was captured using schlieren. The results document the fluid dynamic interactions such as the evolution of a strong shock wave emanating from the strap-on nozzle and the ensuing vortex ring, their interaction with the core jet flow, etc., using flow visualizations and acoustic measurements.

1:25

1pNS2. Flow and acoustics of an isothermal supersonic 2:1 rectangular jet with an adjacent surface. Ephraim Gutmark, Florian Baier, and Aatresh Karnam (Aerosp. Eng., Univ. of Cincinnati, Rhodes Hall 799, Cincinnati, OH 45221-0070, ephraim.gutmark@uc.edu)

In integrated airframe/propulsion system configurations, additional noise sources can be created from the interactions between the jet flow and the adjacent airframe surfaces. Another situation of surfaces of close proximity occur during takeoff and landing when the ground is close enough to cause the jet-surface interference. The impact of a plate oriented parallel to the axis of a 2:1 rectangular supersonic jet ($D_e = 20.65$mm) at the minor axis side is studied. Plate offset (h) distances of h = 0, 1, 2, and 3 equivalent diameters from the nozzle lip to the surface are included. The impact of the plate is studied at nozzle pressure ratios (NPRs) of 2.5–4.5 when the design Mach number is 3.67, at a temperature ratio of TR = 1.1. Mean and turbulent flow field data are acquired using streamwise particle image velocimetry (PIV). Trends of shock cell spacing, potential core length, and shear layer development are analyzed. Near and far field data are taken and correlated with the flow field details. The offset from the nozzle is shown to vary flow and acoustic properties and impact screech tones.

1:45

1pNS3. Laser optical measurement of acoustic phenomena in the near field of a supersonic jet using 2-D position sensitive detector. Koji Okamoto, Kazuya Fukatsu, Masahito Akamine (Dept. of Adv. Energy, Graduate School of Frontier Sci., The Univ. of Tokyo, 5-1-5, Kashiwanoha, Kashiwa, Chiba 277-8561, Japan, k-okamoto@edu.k.u-tokyo.ac.jp), and Susumu Teramoto (Dept. of Aeronautics and Astronautics, Graduate School of Eng., The Univ. of Tokyo, Bunkyo, Tokyo, Japan)

Acoustic measurement close to noise sources is significant to understand the generation mechanisms of jet noise. Microphones are generally used for the acoustic measurement, but they may disturb the flow and acoustic fields when they are used in the near field of a jet. In this study, an optical measurement method using a laser and 2-D position sensitive detector (PSD) is proposed for the acoustic measurement in the near field. In this method, 2-D PSD detects the angle and direction of the refraction of the laser path by acoustic wave passing, so that the propagating direction, as well as the acoustic intensity, is expected to be measured by this method. To discuss its validity, this method was applied to the acoustic measurement of a correctly expanded supersonic jet. In this experiment, microphone measurement was also carried out simultaneously, and cross correlation between the signals of these two measurements is discussed. Also, the measured spectra and propagating directions for different frequencies are compared with those of the acoustic intensity vector measurement.

2:05

1pNS4. Experimental investigation of the impacts of total temperature non-uniformities on the flow and acoustic fields of a heated supersonic jet. Kyle A. Daniel (Aerosp. and Ocean Eng., Virginia Tech, 460 Old Turner St., Blacksburg, VA 24060, kyled1@vt.edu), David Mayo (Mech. Eng., Virginia Tech, Blacksburg, VA), Kevin T. Lowe (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA), and Wing Ng (Mech. Eng., Virginia Tech, Blacksburg, VA)

In recent years, the noise produced by tactical aircraft has become a growing concern due to stricter community noise standards and the negative health effects it has on flight support personnel. This study proposes the examination of a new novel noise reduction method involving thermal non-uniformities in heated supersonic jets. Thermal non-uniformities have the advantage of increasing turbulent mixing in jets without the use of additional hardware and can most likely be implemented in afterburning engines with minimal modification. In the course of this study a thermally non-uniform Mach 1.5 heated jet with a centered and offset thermal non-uniformities will be examined. It will be shown that thermal non-uniformities reduce the far-field radiated noise up to ~2dB at peak frequencies and have a measurable impact on the directivity of radiated noise, including a narrowing of the Mach wave emission angle and a non-uniform azimuthal directivity. These changes in the acoustic field are directly related to global changes in the turbulence development observed in the jet. These effects include a shortened potential core, increased shear layer thickness, and a decreased mean flow in regions of peak Reynolds shear stress. These effects are captured by stereoscopic PIV and near and far-field microphone measurements.

2:25

1pNS5. Visualization of acoustic phenomena of imperfectly expanded supersonic jets using acoustic-triggered conditional sampling analysis. Yuya Sekiguchi, Koji Okamoto (Graduate School of Frontier Sci., Univ. of Tokyo, Kashiwanoha 5-1-5, Kashiwa, Chiba 277-8561, Japan, 9601548985@edu.k.u-tokyo.ac.jp), and Susumu Teramoto (Graduate School of Eng., Univ. of Tokyo, Bunkyo, Tokyo, Japan)

For rockets and supersonic aircrafts, acoustic waves from an exhaust jet cause problems of vibration and noise. To reduce the influence of acoustic waves effectively, it is important to reveal characteristics of acoustic phenomena, such as source locations and the relation between the flow and acoustic phenomena. For this purpose, the authors carried out visualization of acoustic field of a correctly expanded supersonic jet in the previous studies, and proposed the acoustic-triggered conditional sampling analysis to obtain acoustic information from high speed schlieren movies. This method is also expected to be applied to other jet acoustic phenomena. In this study, this method is applied to imperfectly expanded supersonic jets and the relation between the flow and acoustic phenomena will be discussed.

2:45–3:00 Break

3:00

1pNS6. Design and evaluation of a high-speed aeroacoustic wind tunnel. Joana Rocha (Dept. of Mech. and Aerosp. Eng., Carleton Univ., 1125 Colonel By Dr., Mackenzie Building, ME 3135, Ottawa, ON K1S 5B6, Canada, Joana.Rocha@carleton.ca)

The high-speed aeroacoustic wind tunnel (HSAWT) at Carleton University is a new facility commissioned with the purpose of facilitating experimental studies of turbulent boundary layer (TBL) induced surface pressure fluctuations. This research is primarily intended for applications related to aircraft cabin noise generation from structures exposed to high-speed flow. This open-jet, blowdown wind tunnel is a unique facility in Canada and one of a few aeroacoustic wind tunnels in the world capable of achieving speeds up to Mach 0.8. Flow is delivered from a nozzle with dimensions of 6.1 cm $\times$ 15 cm to a test section enclosed within an anechoic chamber with internal dimensions of 1.9 m $\times$ 0.88 m $\times$ 1.95 m. This study details the complete design methodology for all major wind tunnel components, including the numerical simulations performed in the validation of the designed components. Results of preliminary test section flow characterization and chamber background noise measurements are discussed. Finally, experimental results of the TBL surface pressure fluctuation spectral behavior developed over a flat test section plate are compared with established data and empirical models available in literature.

Contributed Papers

3:20

1pNS7. Resolvent analysis for jet noise source identification. Ethan M. Pickering, Georgios Rigas, Oliver T. Schmidt, and Tim Colonius (Mech. Eng., California Inst. of Technol., 1200 East California Boulevard, Mail Code MC 104-44, Pasadena, CA 91125, pickering@caltech.edu)

We use resolvent analysis and spectral proper orthogonal decomposition (SPOD) to deduce the acoustic sources for an isothermal Mach 1.5 round jet. Both physics-based resolvent analysis and data-driven SPOD (using a high-fidelity, experimentally-verified, large-eddy simulation (LES) database) provide a basis for predicting the perturbation field. Singular value decomposition of the resolvent operator based upon the LES baseflow provides optimal volumetric forcing modes, or sources, and their associated linear responses. To identify physically relevant resolvent modes, comparisons are made between the highest gain responses and the highest energy SPOD modes computed directly from LES realizations. The prevalence of the associated resolvent forcing modes in the data are then assessed by projecting them onto the full LES nonlinear terms. The resulting distributions are presented and a jet noise model leveraging these forcing statistics is discussed. [This research was supported by a grant from the Office of Naval Research (Grant No. N00014- 16-1-2445). Ethan Pickering was supported by the Department of Defense (DoD) through the National Defense Science & Engineering Graduate Fellowship (NDSEG) Program.]

3:35

1pNS8. Fundamental study on the acoustic fields caused by the interference of supersonic jet with a perforated plate. Yo Murata, Kazutoki Iwasa (The Univ. of Tokyo, 7-3-1, Hongo Bunkyo-ku, Tokyo 113-8656, Japan, ymurata@fiv.t.u-tokyo.ac.jp), Tatsuya Ishii (Japan Aerosp. Exploration Agency, Tokyo, Japan), Koji Okamoto (The Univ. of Tokyo, Kashiwa, Chiba, Japan), and SHIGEHIKO KANEKO (The Univ. of Tokyo, Tokyo, Japan)

The supersonic jet contains intensive sound sources. It is known that when this jet collides with the perforated plate, a complicated flow field is formed on the plate and a high-amplitude sound is radiated as compared with the case without the perforated plate. Attention is paid to understanding the sound generating mechanism and proposing the measures to alleviate the sound pressure levels. In this study, two types of fundamental model tests have been carried out with a simple combination of a nozzle and a perforated plate. One is acoustic measurement in an anechoic facility. Test results indicated the amplified sound pressure levels in the forward arc of the nozzle. The aperture geometry suggested certain suppression in sound pressure levels in the arc. The other test is an optical visualization with Schlieren photographs. The photographs tried to account for reflection of high-amplitude sound waves on the plate downstream the nozzle.

3:50

1pNS9. Coherence analysis of the simulated sound field of a highly-heated laboratory-scale jet. Kevin M. Leete, Kent L. Gee (Brigham Young Univ., Brigham Young University, Provo, UT 84604, kevinmatthewleete@gmail.com), Junhui Liu (Naval Res. Lab., Washington, DC), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and S. Hales Swift (Brigham Young Univ., Provo, UT)

Large eddy simulations (LES) have successfully reproduced the flow and acoustic fields generated by laboratory-scale jets. However, laboratory-scale test conditions ordinarily do not match those of full-scale military aircraft. Recently, Liu *et al.* (AIAA 2016–2125) showed that accounting for a variable ratio of specific heats leads to increased accuracy in LES simulations of heated jets. As a step towards modeling operating parameters similar to military aircraft, they produced a simulation of a highly-heated jet with a temperature ratio of seven. This paper shows the levels in the field as well as the axial and azimuthal coherence trends of this simulated jet. Large axial coherence lengths are found in the direction of maximum radiation with decreased coherence towards the sideline. Azimuthally, coherence length scales are greatest in the region of maximum radiation and generally decrease with increasing frequency. Additionally, evidence is seen for multiple self-coherent and mutually-incoherent radiation lobes within the region of maximum radiation. [Work supported by USAFRL through ORISE.]

4:05

1pNS10. Acoustic/drag interaction and its impact on future aircraft design. Christopher Jasinski (Mech. Eng., Univ. of Hartford, 54162 Ironwood Rd., South Bend, IN 46635, chrismjasinski@gmail.com)

For several decades, acoustic liner technology has substantially reduced the noise created by commercial jet engines. Conventional placement of the perforate-over-honeycomb liner is in the fore and aft bypass duct of an engine nacelle; however, future aircraft design may incorporate liners on additional surfaces, such as the underside of wings and the fuselage. In order to use liner technology in novel locations, the impact of the porous facesheet and core of the liner on the aerodynamic drag of the craft must be fully understood. Work conducted at the University of Notre Dame has experimentally evaluated this drag, as well as the impact of acoustic fields on the aerodynamic drag produced by acoustic liners. Measurements of several quantities across the full boundary layer profile show this impact is substantial and quantifiable. Such results for several liner facesheets will be presented. Through further testing of the acoustic impedance of various facesheets at the University of Hartford, the goal of striking design balance between acoustic performance and aerodynamic reliability can become more clear. Fluid mechanic, acoustic, and direct aerodynamic measurements for several liner samples will be presented and discussed.

4:20

1pNS11. Comparison between predicted and measured Stevens perceived level of sonic booms for community response testing. William Doebler (NASA Langley Res. Ctr., M.S. 462, NASA Langley Res. Ctr., Hampton, VA 23681, william.j.doebler@nasa.gov)

In 2011, NASA conducted a flight test to study community response and perception of low amplitude sonic booms in the Waveforms and Sonic Boom Perception and Response (WSPR) experiment. Several noise monitors were placed in a 1.6 sq. km area to record the booms. In future sonic boom community response flight tests, it will not be practicable to place noise monitors at the locations of every survey respondent. Therefore, noise predictions are necessary. Using NASA's PCBoom code, the Stevens Mark VII Perceived Level (PL) was predicted at each noise monitor locations using the real aircraft trajectories and weather balloon data taken during the WSPR test. These predictions are then compared to the measured PL values. This comparison will inform the methodology for estimating noise exposure in future flight tests.

4:35

1pNS12. Updated evaluation of sonic boom noise metrics. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov), Sara R. Wilson (Systems Eng. and Eng. Methods Branch, NASA Langley Res. Ctr., Hampton, VA), and Jonathan Rathsam (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

There is no internationally agreed-upon standard noise metric that can be used to quantify sonic boom levels from overflight of supersonic aircraft. Several laboratory studies have investigated perception of sonic booms in outdoor environments, as well as in indoor environments, where transmission loss alters the spectral content and level of the sonic boom and additional factors such as secondary rattle noise and vibration affect perception. Each study has previously been evaluated separately for performance of noise metrics, and the results do not clearly indicate a preferred metric. Meta-analyses have also been performed to evaluate the performance of metrics across five of these studies. An additional sixth study of human response that incorporates rattle noise resulting from sonic booms at representative levels was recently conducted and has been added to the meta-analysis. The analysis considered thirteen metrics with three meta-analysis methodologies: individual subject ratings for indoor studies, mean ratings for indoor studies, and mean ratings for both indoor and outdoor studies. Considering both indoor and outdoor studies, this latest analysis confirmed previous results on the five metrics with highest correlation (r^2) across studies: PL, BSEL, DSEL, ESEL, and ISBAP (Indoor Sonic Boom Annoyance Predictor).

Session 1pPA

Physical Acoustics, Noise, and ASA Committee on Standards: Outdoor Sound Propagation II

Vladimir E. Ostashev, Cochair
U.S. Army Engineer Research and Developmnet Center, 72 Lyme Rd., Hanover, NH 03755

Philippe Blanc-Benon, Cochair
Centre acoustique, LMFA UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290

Contributed Papers

3:30

1pPA1. Frequency dependent atmospheric scattering of infrasound: Observations and modeling of controlled explosions. Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., Univ. of California at San Diego, 9500 Gilman Dr., La Jolla, CA 92037-0225, chedlin@ucsd.edu)

In deriving equations relating to the amplitude of infrasound signals from large explosions to the energy of the source, the assumption is generally made that the amplitude scales linearly with the source yield. However, results from experiments with large-yield detonations carried out at the Utah Test and Training Range (UTTR) west of Salt Lake City have cast doubt on that assumption, at least within the near field. In 2016, acoustic and infrasound sensors were placed at ranges up to 90 km east of the UTTR test site, and sources with yield ranging from 1700–17700 kg were detonated during the summertime. In several cases, detonations occurred only several hours apart. Predictable changes in amplitude might be expected, given that only small-scale spatial variations in the atmosphere occur over that time. The fact that this is not observed may be attributable to frequency dependent effects in atmospheric scattering. I report on the observations and show the degree to which amplitude variations can be predicted with numerical modeling. Numerical computations of the Navier-Stokes equations governing acoustic propagation are performed to investigate infrasound propagation for these events. The modeling allows for nonlinear propagation within an azimuthally symmetric atmospheric model, and incorporates accurate weather information.

3:45

1pPA2. Estimation of atmospheric sound transmission loss due to sea roughness by mean of an effective impedance: A numerical validation. 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 (Dept. of Eng., East Carolina Univ., Greenville, NC), and Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, Washington, DC)

Preliminary numerical studies have been demonstrated that sea roughness introduces an excess of transmission loss of atmospheric sound propagation. However, numerical solution of the parabolic equation that describes this phenomenon is computationally intensive, particularly when the spatial domain is large (>1 km). This study evaluates the validity of implementing an effective impedance (a flat absorbing surface) in the place of a rough surface. The effective impedance is a parametric complex quantity that is different for each sea state. The parameters of the impedance are estimated by best fitting the numerical solution of a two-dimensional finite-difference time-domain solver. The effectiveness and validity of using this impedance in place of a rough surface is evaluated by direct comparison of the solutions of the Crank-Nicolson parabolic equation obtained with and without the impedance.

4:00

1pPA3. Sea surface contribution to sound transmission loss: Numerical comparison between three-dimensional and cylindrical symmetry approximation solutions. Andrea Vecchiotti, Luca Gasparini (Mech. Eng., Università Politecnica delle Marche, Senigallia (AN), Italy), Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), Joseph F. Vignola, and Diego Turo (Mech. Engeneering, The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, turo@cua.edu)

The atmospheric sound propagation over the sea depends on several factors including but not limited to temperature, wind speed, relative humidity, and roughness of the surface. The objective of this study is to determine the contribution to sound transmission loss of the sea surface roughness. The governing equations are implemented for a three-dimensional domain using a finite element method solver. The sea surface is assumed to be perfectly reflective and its shape is pseudo-randomly generated by using the Pierson and Moskowitz model. In previous studies, the cylindrical symmetry approximation has been widely used to allow a fast computation of the equations. However, the implementation of a three-dimensional domain allows a more realistic scenario and is here used as a tool to quantify the deviation from the approximated solution. An appropriate correction factor to be used when applying the cylindrical symmetry approximation is determined.

4:15

1pPA4. Experimental characterization of atmospheric boundary layer humidity profile. Teresa J. Ryan (Dept. of Eng., East Carolina Univ.,248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Joseph F. Vignola, John Judge, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents a measurement system designed to capture the humidity profile in the near-surface (up to 100 m) atmospheric boundary layer. The sensor array combines a portable static array of humidity sensors as well as GPS-synchronized pressure, temperature, and humidity sensors mounted to an unmanned aerial vehicle. These humidity measurements support an ongoing effort that is investigating the influence of humidity profile on acoustic ducting phenomena. These measurements will inform sensitivity

studies using a numerical model being developed in a parallel effort. Preliminary measurements are presented for a range of experimental conditions such as foreshore or littoral, freshwater wetland, and dry coastal plain. The aim of this overall effort is an improved numerical model of acoustic refraction and attenuation over moderate to long distances above a sea surface that accounts for sea state as well as boundary layer wind, temperature, and humidity profiles.

4:30–4:45 Break

4:45

1pPA5. On atmospheric humidity and acoustic ducts. Joseph F. Vignola, Diego Turo, John Judge (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), and Teresa J. Ryan (Engineering, East Carolina Univ., Greenville, NC)

The development of various propagation ducting phenomena (elevated or surface) has been studied extensively for electromagnetic and electro-optical wave propagation. The influence of changes in temperature, pressure, and humidity gradients on the strength and geometry of the duct in the electromagnetic propagation problem is well-documented, and there is substantial literature regarding the influence of temperature gradient on atmospheric acoustic propagation. Analytical results indicate that moisture gradients in warm air above a sea surface contribute meaningfully to the refraction of sound. This work uses numerical simulation to explore the atmospheric conditions that would support development of acoustic ducts or wave guide conditions during moderate range ($<$1 km) acoustic propagation. The aim of this overall effort is an improved numerical model of acoustic refraction and attenuation over moderate to long distances above a sea surface that accounts for sea state as well as boundary layer wind, temperature, and humidity profiles.

5:00

1pPA6. Measurements of the farfield characteristics of an automotive ultrasonic sensor with a vertical temperature gradient. Sang Hyun Kim, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, ares3833@naver.com), and Youngsoo Choi (Hyundai MOBIS, Yongin-si, South Korea)

With the coming of autonomous vehicles on the horizon, a variety of sensor platforms such as ultrasonic, radar, lidar, and image sensors are currently under development. Among these, ultrasonic sensors play a key role in the autonomous parking system. An ultrasonic sensor emits and receives a sound pulse (usually centered around a few dozen kHz), and computes the distance to a nearby object assuming a constant sound speed. However, the assumption of constant sound speed is often invalid because of the meteorological conditions, and could cause errors in distance estimation. In this talk, we discuss the farfield characteristics of a typical automotive ultrasonic sensor in the presence of a vertical stratification of temperature hence sound speed. The sound speed profile on a hot summer day is derived from temperature measurements with a vertical thermocouple array, and the farfield acoustic pressure is measured using a 2-D array of microphones. The measurements indicate that refraction due to the stratification of sound speed can be significant enough to alter the farfield performance of an ultrasonic sensor, and thus should be taken into account in the design of the autonomous parking system.

5:15

1pPA7. Experimental study on aerodynamic noise characteristics of high-lift configuration with variable gap leading-edge slat. Weishuang Lu, Peiqing Liu, and Hao Guo (Key Lab. of Aero-Acoust. (Beihang University), Ministry of Industry and Information Technol., Xueyuan Rd. No.37, Haidian District, Beijing 100191, China, lujiaww@163.com)

The aircraft high-lift devices are important components to ensure the safety of take-off and landing, and it is also the main source of airframe noise in the process of take-off and landing, especially the leading-edge slat. The slat that reduces gap by rotating with the fixed axis is proposed in this paper, and experimental study of far-field aerodynamic noise characteristics of high-lift configuration with the slat is also carried out. The experimental results show that discrete tones in the low-middle frequency range disappear and the broadband frequency amplitude also slightly decline, resulting in a significant decrease in the overall sound pressure level of the high-lift configuration with slat gap of 0, compared with the configuration with conventional leading-edge slat. However, in the process of rotation from conventional slat to seamless slat, more prominent discrete tone in low-middle frequency range or humps in mid-high frequency range appear at angles of attack below 6°, which give rise to the overall increase in the sound pressure level. At angles of attack above 6°, these aero-acoustic phenomena are not obvious. Hence, the overall sound pressure levels of high-lift configuration gradually decrease with the rotation process in larger angles of attack.

Session 1pPP

Psychological and Physiological Acoustics: Understanding Limitations on Auditory Spatial Acuity

Andrew D. Brown, Chair
Physiology & Biophysics, University of Colorado School of Medicine, 12800 East 19th Avenue, RC1-N, Rm 7401G, Mail Stop 8307, Aurora, CO

Chair's Introduction—2:00

Invited Papers

2:05

1pPP1. Sense, the single neuron, and multi-dimensional stimuli. Victor Benichoux (Institut Pasteur, 25 rue du Dr Roux, Paris 75015, France, victor.benichoux@pasteur.fr)

Sensory processing disorders are characterized by poor behavioral performance in psychophysical tasks relative to the general population. To assess the etiology of those deficits, a tempting possibility is to look for abnormalities in the information available in responses of specific groups of neurons, which could then become the target of rehabilitation. I discuss here basic limitations in linking behavioral performance to neuronal responses. Our understanding of sensory neurons comes from measuring *tuning curves,* which describe how the response varies when a dimension of the stimulus is varied. In reality, however, neural responses are modulated by many stimulus features, e.g., in the auditory system: amplitude and spectrum of the stimulus, spatial position, temporal properties, etc. I illustrate the challenges that this tuning heterogeneity raises when trying to predict results of behavioral experiments from neuronal responses. I first exhibit effects in theory using Fisher Information, then analyze explicit optimal decoders trained to estimate spectral and spatial properties of an auditory stimulus from neural responses. I show how heterogeneity in tuning is a major contributor to the robustness of the encoding of multiple stimulus dimensions. Finally, I described how this changes the interpretation of neural constraints on behavioral acuity.

2:25

1pPP2. Changes in the minimum audible angle may be related to systematic distortions in sound localization and motion perception. W. Owen Brimijoin (Facebook Reality Labs, MRC/CSO Inst. of Hearing Res., Fl. 3, New Lister Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, owen.brimijoin@nottingham.ac.uk)

Systematic distortions in our perception of acoustic space have been repeatedly documented in the literature. For example, listeners exhibit a characteristic overestimation of azimuth when they are asked to point a laser at a sound source while keeping their head and eyes fixated ahead of them. Motion perception also appears to be non-uniform as a function of azimuth, as demonstrated by the fact that listeners need roughly twice as much motion at the side than at the front to judge the two motions as equivalent. This talk will make the case for a common source of these two perceptual inconsistencies, namely, the change in minimum audible angle (MAA) as a function of azimuth. Classically speaking, the MAA is thought of as a threshold measure that increases at more eccentric angles because of a change in the availability of binaural cues. Regardless of the source, we argue here à la Fechner that azimuth overestimation and motion perception non-uniformity are suprathreshold consequences of the change in acuity. By treating, e.g., 20° of acoustic space at the side as equivalent to 10° at the front, a model of the perceptual topology of acoustic space can capture several previously reported findings in spatial hearing.

Contributed Paper

2:45

1pPP3. Time-series analysis of binaural perception and uncertainty using eye movements. Matthew Winn (Speech Lang. & Hearing Sci., Univ. of Minnesota, 164 Pillsbury Ct, Minneapolis, MN 55455, mwinn83@gmail.com) and Gabrielle O'Brien (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Uncertainty affects sensory perceptions, even for stimuli well above detection thresholds. Using an anticipatory eye-movement paradigm popularized in studies of linguistic perceptual certainty and competition, we measured perception of binaural cues in dichotic narrowband noises that varied by center frequency, bandwidth, envelope fluctuations and interaural correlation. Participant's eye gaze served to mark detection of change in the ILD or ITD. We tracked the timing and direction of saccades, as well as the number of saccades ("guesses") as a metric of perceptual uncertainty. Results showed gradually increasing performance accuracy for greater cue magnitude, coupled with fewer saccades overall. There was also a change in the distribution of saccade timings and number of saccades depending on whether cues were low-frequency ITD, high-frequency envelope ITD, correlated-noise ILD, or uncorrelated-noise ILD. All ILD results were consistent with a model of detection based on interaural envelope decorrelation; performance was particularly poorer, slower, and more variable for stimuli whose envelopes had greater fluctuation, but only when channels were uncorrelated. Comparison with behavioral accuracy and reaction time will be discussed. This study highlights the potential benefits of time-series analysis in psychoacoustic experiments in order to reveal some subtle effects of stimulus properties on perceptual judgments and certainty.

3:00

1pPP4. Effects of interaural delay, center frequency, and no more than "slight" hearing loss on binaural processing: Behavioral data and quantitative analyses. Leslie R. Bernstein and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, University of Connecticut Health Ctr., Farmington, CT 06032, lbernstein@uchc.edu)

Recent neurophysiological studies concerning "hidden hearing loss" have motivated investigators to attempt to uncover behaviorally-measured auditory deficits that might exist in human listeners who would be categorized via standard audiometry as having normal hearing or, at most, "slight" hearing loss. We hypothesized that behavioral measures of binaural auditory performance could reveal the presence of deficits that may not be discoverable via standard audiometric testing or via measures of monaural processing. That hypothesis was based on the notion that monaural (perhaps, peripheral) deficits would be effectively multiplied via binaural interaction. Groups of listeners, all having no greater than "slight" hearing loss were tested. Binaural detection thresholds were elevated across a wide range of center frequencies for listeners whose absolute thresholds at 4 kHz exceed 7.5 dB HL. Quantitative predictions made via an interaural correlation-based model of binaural processing suggest that the elevated binaural detection thresholds observed for listeners having slightly elevated absolute thresholds stem solely from their having elevated levels of internal noise. They appear to stem neither from reduced sensitivity to signal-dependent changes in information derived from the stimuli as processed internally nor from increased noise along the listener's putative internal delay line.

3:20–3:35 Break

3:35

1pPP5. Behavioral and neural spatial acuity can be persistently reduced by early temporary hearing loss. Daniel J. Tollin and Kelsey Anbuhl (Physiol. & Biophys., Univ. of Colorado School of Medicine, University of Colorado Anschutz Medical Campus, 12631 E 17th Ave., Aurora, CO 80045, Daniel.Tollin@ucdenver.edu)

Children experiencing asymmetrical hearing early in life, typically due to conductive hearing loss (CHL) associated with ear infection, often display reduced behavioral spatial acuity that persists long after hearing is restored, suggesting abnormal central auditory development. We hypothesized that persistent CHL disrupts the experience-dependent fine-tuning of binaural hearing necessary in the developing auditory system to support normal behavioral spatial acuity. Using an animal model (the guinea pig), we found that chronic unilateral CHL during development (induced by an earplug) caused the brain to generate a less precise representation of auditory space. When the hearing loss was reversed by simply removing the earplug, both the spatial acuity of single neurons in the inferior colliculus and behavioral performance in a simple sound location acuity task were ~threefold worse than animals that had not worn earplugs, as if the sense of auditory space had been blurred. Overall, the results suggest that experiencing even temporary hearing loss during early development can persistently alter the normal maturation of the auditory brain circuits that are necessary to support good spatial acuity. Thus, the maladaptive plasticity in these circuits due to temporary hearing loss during development can place limitations on spatial acuity in adulthood. [Support: R01-DC011555, T32-DC012280, F31-DC014219.]

3:55

1pPP6. Does electrode placement affect the interaural-time-difference acuity in bilateral cochlear-implant listeners? Olga A. Stakhovskaya (Hearing and Speech Sci., Univ. of Maryland, College Park, 4954 North Palmer Rd., Bldg. 19, R. 5607, Bethesda, MD 20889, olga.stakhovskaya.ctr@mail.mil), Joshua G. Bernstein (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Jack H. Noble (Dept. of EECS, Vanderbilt Univ., Nashville, TN), Kenneth K. Jensen (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., Eden Prairie, MN), Michael Hoa, Hung J. Kim (Dept. of Otolaryngology-Head and Neck Surgery, Georgetown Univ. Medical Ctr., Washington, DC), and Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD)

Postlingually deafened bilateral cochlear-implant (CI) listeners can show limited interaural-time-difference (ITD) sensitivity, even when tested using highly controlled time-synchronized research processors. It is assumed that ITD discrimination requires interaural frequency-matched inputs. However, current bilateral CI programming procedures do not account for potential interaural place-of-stimulation mismatch. This study investigated the magnitude of interaural place-of-stimulation mismatch and its effects on ITD sensitivity in bilateral CI listeners. Ten bilateral CI listeners were tested on a two-interval left-right ITD discrimination task. Loudness balanced, 300-ms, 100 or 200 pulse-per-second, constant-amplitude, monopolar pulse trains were delivered to single-electrode pairs using time-synchronized research processors. ITD just noticeable differences (JNDs) were measured for five reference electrodes evenly distributed along the array in one ear, and for at least five comparison electrodes, generating ITD tuning curves. The interaural mismatch was estimated using both the ITD tuning curves and differences in the angular insertion depth from computed-tomography (CT) scans. Data showed that ITD tuning curves were relatively broad when compared to the amount of physical interaural place-of-stimulation mismatch from the CT scans. This suggests that most bilateral CI listeners do not experience appreciably reduced binaural sensitivity due to differences in electrode placement.

Contributed Paper

4:15

1pPP7. Lateralization of interaural time and level differences measured with cochlear implant sound processors. Alan Kan and Ruth Litovsky (Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

Recently, our lab developed a novel technique that enables binaural sensitivity to be measured via unsynchronized clinical cochlear implant (CI) sound processors. By using a transposed-tone complex with carrier frequencies at the center frequency of the patient MAP, bilateral CI users have measurable sensitivity to interaural time and level differences (ITD and ILD, respectively) that is on par with results from bilaterally synchronized research platforms. In this work, we measure the extent to which a transposed-tone complex presented via clinical processors can be mapped to lateral locations in the head as a function of changes in ITD and ILD. In addition, lateralization was measured when ITDs and ILDs co-varied. Results show that listeners are able to lateralize the tone complex throughout the range of the head. However, the slope of the ITD lateralization function appears to be shallower than that of ILDs. When ITDs and ILDs co-vary, the slope of the lateralization function appears to follow that of ILDs, suggesting that bilateral CI users may not be combining the two cues for sound source lateralization.

Invited Paper

4:30

1pPP8. Exploiting spatial audio cues for pilot navigation in degraded visual environments. Heath Jones, Lana Milam, Stephanie Karch (APPD, USAARL, Bldg. 6901, Fort Rucker, AL 36362, heath.g.jones2.ctr@mail.mil), Henry Williams (Naval Medical Res. Unit Dayton, Dayton, OH), and Brian Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Aviation mishaps resulting from degraded visual environments (DVEs) represent a significant loss in military personnel and aircraft every year. DVEs are considered any type of environmental condition (e.g., sand, snow, or fog) that visually obstructs the pilot and can cause spatial disorientation. The current study represents a tri-service effort exploring the implementation of spatial audio cueing techniques to aid pilot navigation in DVEs. Directional cueing (i.e., indicating the location of target waypoints) was achieved by spatializing an auditory stimulus using the SoundLab audio rendering package and convolving audio signals with (non-individualized) head-related transfer functions. Two spatial cue conditions were tested, either rendered dynamically in reference to the pilot's head via head tracking or with respect to aircraft heading. Data were collected from pilots operating a full-motion UH-60 Black Hawk flight simulator at the U.S. Army Aeromedical Research Laboratory. Pilots performed multiple flight maneuvers, such as a "turn to target" localization task, a side-step maneuver positioning the aircraft over a stationary target, and an "approach to moving target" task. Performance was assessed by measures of localization error, completion time, and failure rate. Findings from this study provide information on sensory cueing display countermeasures for helicopter flight in DVEs.

Session 1pSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Advances in Thermoacoustics

Matthew Kamrath, Cochair
Signature Physics Branch, US Army ERDC-CRREL, 72 Lyme Road, Hanover, NH 03755

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

Invited Papers

1:00

1pSA1. Thermoacoustic waste heat recovery engine. Comparison of simulation and experiment. Thomas W. Steiner and Maarten Elferink (Etalim, 62 West 8th Ave., Ste. 400, Vancouver, BC V5Y-1M7, Canada, tsteiner@etalim.com)

A thermoacoustic engine heated with a 10–20 kW 600C waste heat exhaust stream was designed and built. The heat is extracted from the exhaust with a microchannel counter flow heat exchanger and delivered to the hot side of the regenerator with a self-ciculating loop. This loop is driven using Venturi driven streaming of the helium working gas so that no high pressure and high temperature pump is required. The prototype system performed close to simulated expectations. A peak of 570 W electrical power was measured as was a peak conversion efficiency of waste heat enthalpy to electrical energy of 5%.

1:20

1pSA2. Monocoque shell and tube heat exchanger for thermoacoustics. Robert M. Keolian, Kevin J. Bastyr, Ray S. Wakeland, and John F. Brady (Graduate Program in Acoust., The Penn State Univ., 732 Holmes St., State College, PA 16803, keolian@psu.edu)

Heat exchangers can be a limiting factor to thermoacoustic power density. They can also be difficult to construct because of a tendency toward many small parts due to the small size of the thermal penetration depth. We present a heat exchanger that consists of around five pieces and yet has very small feature size and is easy to construct in large numbers. A sacrificial mandrel, consisting of a sheet with a lattice of specially shaped holes, is formed by casting in a mold with a lattice of specially shaped pins. The mandrel is plated with nickel, electroless nickel, or other metal. The mandrel is then removed by melting or dissolving, leaving a monocoque tube bundle. A screen mesh may be joined to one or both faces of the monocoque to act as fins to improve thermal contact to the working fluid that flows through the tubes. Coolant flows around the tubes within the monocoque shell. For a typical device, we find that without a mesh on the monocoque, lost work is minimized with tubes on the order of 8 mm long and spaced by 0.4 mm, but with a 100-mesh copper screen on both faces, the tubes can be a more manageable 3 mm long spaced by 4 mm.

1:40

1pSA3. Standing-wave and traveling-wave thermoacoustics in solid media. Haitian Hao, Carlo Scalo (Mech. Eng., Purdue Univ., Herrick Labs, 177 S. Russell St., Rm. 1007, West Lafayette, IN 47906, haoh@purdue.edu), Mihir Sen (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN), and Fabio Semperlotti (Mech. Eng., Purdue Univ., West Lafayette, IN)

Thermoacoustics is a multi-physics process in which heat and mechanical power (associated with waves) can be converted into one other. This process has been exploited to design different types of modern-day energy conversion devices, such as thermoacoustic engines (TAE) and refrigerators (Swift, 1988), since the first experimental assessment of the phenomenon by Soundhauss (1850) in the glass blowing process. To date, all the thermoacoustic devices are fluid-based, using mostly air or Helium as the working medium. Our study explores for the first time the possibility of achieving thermoacoustic energy conversion in solid media, by laying out the fundamental theory and showing numerical evidence of the existence of both standing and traveling thermoacoustic waves in solids. Consistent with established results for fluid-based TAEs, the growth-rate-to-frequency ratio of traveling waves in solids is found to be significantly higher than that of standing waves. While sold-state thermoacoustics (SSTA) share some commonalities with their fluid counterpart, some important and noteworthy distinctions are present. For example, solids have the potential to be highly engineered (e.g., metamaterials), with their properties tuned to enhance thermoacoustic energy conversions. The theoretical investigation of this mechanism may motivate novel ideas designing a new generations of ultra-compact, highly efficient thermoacoustic devices.

2:00

1pSA4. Study on a coaxial thermoacoustic system: Effect of the edge shape on the acoustic intensity. Yukihiro Takeyama (Elec. and Electron. Engineering, Doshisha Univ., 1-3 Tatara-miyakodani, Kyotanabe-city, Kyoto, Kyotanabe 610-0321, Japan, ctwb0351@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Electron. System Eng., Univ. of Shiga Prefecture, Hikone, Japan), Masaya Saeki (Elec. and Electron. Engineering, Doshisha Univ., Kyoto, Japan), Daichi Kuroki, and Yoshiaki Watanabe (Biomedical Information, Doshisha Univ., Kyotanabe, Japan)

For the downsizing, the coaxial thermoacoustic system (coaxial-type) has been proposed. The coaxial-type is expected to have high efficiency because the energy conversion with traveling wave phase like looped-tube is realized. Additionally, this system is more compact than the looped-tube. The circular flow path of the coaxial-type is made of an annular flow path between the inner and the outer tube and a cylindric flow path in the inner tube. By propagating in the cylindric flow path after passing through the annular flow path, a traveling-wave sound field is generated. The edge space between the annular flow path and the inner tube was studied due to the reduction of the propagation loss. The influence of the inner tube length and the edge shapes of outer tube on the energy conversion was examined. The two edge shapes were used: the flat one and the spherical one. It was confirmed the sound energy generated in the system with the spherical shape is higher than the one with the flat shape. It is suggested a traveling-wave sound field is easily generated by the spherical shape because of the focus effect. Thus, the edge shape is one of important points for designing the coaxial-type.

2:20

1pSA5. Effect of the resonance frequency on the work flow with adjusting the internal temperature of the stack in a standing-wave thermoacoustic-system. Mana Sugimoto (Doshisha Univ., Tatara Miyakodani 1-3, Kyotanabe, Kyoto 610-0321, Japan, ctwb0349@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Pref., Hikone, Shiga, Japan), Yuya Kurata, Yuto Kawashima, and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Kyoto, Japan)

We discuss the stack which is the key device of the thermoacoustic system. We elucidate the stack mechanism to realize further energy conversion efficiency improvement. We propose installing a heater inside the stack. This heater adjusts the internal temperature of the stack and controls the heat flow in the stack. Controlling the heat flow in the stack can also affect the mutual conversion of the heat flow and the work flow. In this report, the effect of the resonance frequency with the standing-wave thermoacoustic-system and the internal temperature of the stack on the heat flow in the stack was experimentally investigated. The resonance frequency was changed with the total length of the system. As a result, the lower the frequency, the larger the amount of the work flow generated in the stack. This can be influenced by the increase and decrease of the dissipation with changing the resonance frequency. Also, at any resonance frequency, the amount of the work flow generation increased with controlling the heat flow in the stack. The heat flow can be controlled by adjusting the internal temperature of the stack at any resonance frequency.

2:40

1pSA6. High power-density thermoacoustic sound projectors. 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)

Design, fabrication, and performance of large size and high power-density encapsulated thermoacoustic sound projectors driven by short pulses will be presented. The analysis of enhanced energy conversion efficiency using proper signal processing and low-dimensional thermodynamics will be given. [Work supported by the Office of Naval Research and Army Research Office.]

3:00–3:15 Break

3:15

1pSA7. Investigations in performance of a gas-combustion infrasound source. Chad M. Smith and Thomas B. Gabrielson (Graduate Program in Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, chad.smith@psu.edu)

Designing a coherent (non-explosive) sound source within the infrasound band (0.1–4 Hz in our case) is a difficult engineering challenge. This is due to the large air mass that must be moved to created useful signal levels. The fundamental simple source equation, which will govern almost any human-made infrasound source due to the long wavelengths, shows this fundamental difficulty. As frequency decreases, the volume velocity must increase by the inverse factor of the frequency decrease in order to maintain an equal pressure amplitude at equal range. In other words, 1000 times the cubic volume of air must be moved at 0.1 Hz as compared to 100 Hz, in order to maintain an equal output pressure amplitude! For this reason, a novel method has been proposed to skirt this engineering challenge; using the large energy density available in gas combustion for periodic thermal expansion of an air mass. This talk builds on previous work by the authors (Smith and Gabrielson, J. Acoust. Soc. Am. **137**, 2407, 2015 and Smith and Gabrielson, J. Acoust. Soc. Am. **143**, 1808, 2018) and presents comparisons of a 1st-order thermodynamic model and empirical measurements using a large liquid-propane burner system from a hot air balloon.

3:35

1pSA8. Modeling thermal effects in the time domain. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

Modeling thermal transfer from fluid to a solid surface is well understood. However, the mathematical description of this behavior includes transcendental functions in the frequency domain. This precludes some kinds of modeling in the time domain. In some simple geometries, however, it has been shown that a continued fraction expansion of the transcendental functions is possible that allows for a finite bandwidth solution in the time domain. This paper will explore the use of these methods for time domain modeling of thermal-acoustic systems.

3:55

1pSA9. Phase diagram of coupled thermoacoustic chaotic oscillators. Rémi A. Delage, Yusuke Takayama, Sato Moeko, Hiroaki Hyodo (Mech. Eng., Tohoku Univ., Adv. Mech. Systems and Design, Graduate School of Eng., Tohoku University 6-6-01, Aoba, Aramaki, Aoba-ku, Sendai 980-8579, Japan, delageremi@hotmail.fr), and Tetsushi Biwa (Mech. Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Linear analysis of thermoacoustic oscillations have been extensively made, but there is still a lack of knowledge on nonlinear phenomena such as synchronization and chaotic oscillations. Thermoacoustic chaotic oscillations have been reported in Taconis and combustion oscillators but these systems do not allow a well-controlled experiment because of the extreme temperature conditions. By making the system acoustically dissonant, we recently succeeded in generating thermoacoustic chaotic oscillations with modest temperatures. In this study, we present spatiotemporal evolution of thermoacoustic chaos from pressure measurements, and report an experimental bifurcation diagram when two thermoacoustic chaotic oscillators are dissipatively coupled to each other. The two-parameter bifurcation diagram is constructed by varying the frequency mismatch and the coupling strength. Complete chaos synchronization is observed in the region with a frequency mismatch of less than 1% of the uncoupled oscillator, with on-off intermittency phenomenon for low coupling strength. In other regions, synchronization between quasiperiodic oscillations and that between limit-cycle oscillations and oscillation death are observed as well as asynchronous states.

Contributed Papers

4:15

1pSA10. Passive structural monitoring based on matched processing. Emma Lubeigt (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8820 Shellback Way, La Jolla, CA 92037, elubeigt@ucsd.edu), Sandrine Rakotonarivo, Serge Mensah, Jean-François Chaix (Aix-Marseille Université, CNRS, Centrale Marseille, Lab. of Mech. and Acoust., Marseille, France), William A. Kuperman, Jit Sarkar (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), François Baqué, and Gilles Gobillot (CEA Cadarache, Saint-Paul-Lez-Durance, France)

The fourth generation of nuclear reactor designs use liquid sodium as the core coolant. Due to both significant primary loads and thermal variations, important supporting structures are susceptible to thermo-mechanical fatigue. Periodic inspections are planned in order to check the health of the immersed structures, particularly to detect or to characterize potentially hazardous damage. The presented approach relies on both the *a priori* knowledge of the inspected structures (geometry and properties) and the low-frequency ambient noise in the reactor vessel. Indeed, the latter is used as a source of opportunity to inspect the immersed elastic structures. Two kinds of potential defects could occur: localized defects (e.g., fatigue cracks) and global defects (e.g., deformation of the structure). The feasibility of using data derived from ambient noise for detecting and localizing these defects is carried out on an academic example: a cylindrical shell immersed in water. A cross-correlation based procedure is used to construct the replica and datasets. Then, techniques, such as Matched Field Processing and Matched Mode Processing, are investigated to detect and locate the damage. Both conventional and adaptive processors are implemented and their performance compared. Presented results are obtained from numerical data.

4:30

1pSA11. Suspension optimization for a hermetic compressor assembled on a refrigerator. Alexandre A. Pescador Sarda (DEMEC, UFPR, Av. Cel. Francisco H. dos Santos, 100, Curitiba, PR 81530000, Brazil, pescador@ufpr.br)

Noise annoyance generated by hermetic compressors is evaluated based on the sound power level (SPL) parameter measured in a reverberation room or using a semi-anechoic chamber. However, when the machine is assembled on a refrigerator, the SPL can be altered depending on the new system configuration and the way the machine is assembled in the final product. The vibration generated in the electrical machine is transmitted to the refrigerator resulting in noise at the surface. The aim of this study was to model the suspension of a compressor assembled in a flexible base and minimize the vibratory power flow transmitted to the base through an optimization process, taking into account parameters such as the spring position. Decreasing the power flow to the refrigerator base results in a reduction in the global levels of noise and vibration at the base plate.

4:45

1pSA12. Artificial Neural Network-DeltaEC Hybrid Model for Thermoacoustic Systems: New Synergistic Approach. Anas M. Abdelrahman and Xiaoqing Zhang (Dept. of Refrigeration and Cryogenics, School of Energy and Power Eng., Huazhong Univ. of Sci. and Technology, 1037#, Luoyu Rd., Hong shan District, Wuhan, China, Wuhan 430074, China, arahman@hust.edu.cn)

Despite the wide-spread use of DeltaEC model in thermoacoustics community, intrinsic nonlinear phenomena are still hindering its applicability to real practical situations due to assumptions and limitations. In the present study, artificial neural network (ANN) as an intelligent technique is hybridized with conventional DeltaEC model to provide a new synergistic approach called (ANN-DeltaEC) hybrid model for thermoacoustic research field. The aim of this paper is to improve the prediction accuracy of single DeltaEC model by integrating it with distributed and synergistic neural networks. One application for this new approach has been conducted on one standing wave thermoacoustic heat engine based on published literature work to predict the acoustic wave parameters, namely, oscillating frequency and acoustic pressure amplitude under given design considerations of stack geometry and resonator length. The results from hybrid synergistic model had been proven to be desirable in its accuracy compared to experimental work and better than the results of DeltaEC model itself. The present work had shown the capability of the new synergistic approach in accurately predicting the outputs for any new given inputs within given range. Further applied research will be devoted in order to identify complex mappings between thermoacoustic system parameters and their corresponding responses.

Session 1pSCa

Speech Communication, Psychological and Physiological Acoustics: Coupling Phonetics and Psycholinguistics I

Ann Bradlow, Cochair
Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL

Yue Wang, Cochair
Linguistics, Simon Fraser University, 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada

Ratree Wayland, Cochair
Linguistics, University of Florida, 2801 SW 81st Street, Gainesville, FL 32608

Chair's Introduction—1:00

Invited Papers

1:05

1pSCa1. Phonetics, psycholinguistics, and language-specificity. Ann Bradlow (Dept. of Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

How does linguistic knowledge influence speech production, perception, and learning? In this presentation, I will review three sources of empirical evidence for language specificity in some of the most basic processes of speech communication. Each line of evidence will highlight the contributions of the remarkable Jongman-Sereno collaboration—the honorees of this special session—and their continuing influence on generations of students. First, we will consider language specific phonetics as it relates to the acoustic details of speech categories across languages with different sound inventories, including Modern Greek, German, English, and Spanish. Second, we will examine the influence of language background on speech perception through comparisons of pitch perception by speakers of languages with and without lexical tone. Finally, we will focus on speech training in adults when presented with novel speech patterns, asking in particular how general learning principles interact with language-specificity for perceptual adaptation to second-language or foreign-accented speech. In each of these domains—production, perception, and learning—we will see the expanding circles of interaction amongst Jongman, Sereno, and their many students and colleagues. Critically, these Jongman-Sereno contributions underscore the importance of cross-language comparison coupled with a highly collaborative research approach for understanding speech processing and learning.

1:25

1pSCa2. Experience, attention, and context in the processing of systematic variation in spoken language. Lynne C. Nygaard (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

The acoustic speech signal is characterized by enormous variability. Characteristics of individual speakers and groups of speakers profoundly change the way in which linguistic structure is realized. A substantial body of research suggests that language users track, retain, and use systematic variation to restructure linguistic representation and processing in order to maximize intelligibility of spoken language. Less clear is whether sources of variation differ in relevance during speech processing and how relevance changes as a function of experience and context. Research will be presented examining talker-, task-, and listener-related factors that mediate memory and learning of systematic variation in spoken language. The findings suggest that although listeners dynamically adapt to systematic changes in linguistic structure as a function of experience, this adaptation depends on the characteristics and frequency of particular sources of variations, the modulation of attention driven by the structure of the learning environment, and expectations and subsequent sensitivity to socially relevant variation. The considerable behavioral and representational plasticity that is characteristic of speech perception and spoken language processing may depend in part on the social, linguistic, and contextual relevance of the variation associated with both individual talkers and classes of talkers.

1:45

1pSCa3. Learning foreign-language sounds in adulthood: Listening, speaking, and individual differences. James M. McQueen, Jana Krutwig (Donders Inst. for Brain, Cognition and Behaviour, Radboud Univ., Montessorilaan 3, 6525 HR, Nijmegen 6525 HR, Netherlands, j.mcqueen@donders.ru.nl), Lisette Jager (Ctr. for Linguist, Leiden Univ., Leiden, Netherlands), Peter Desain (Donders Inst. for Brain, Cognition and Behaviour, Radboud Univ., Nijmegen, Netherlands), Jurriaan Witteman, and Niels 0. Schiller (Ctr. for Linguist, Leiden Univ., Leiden, Netherlands)

Adult native Dutch speakers tend to have difficulty learning the English /æ/-/ɛ/ contrast because both English vowels can be assimilated to Dutch /ɛ/. Two experiments examined how this contrast is acquired by Dutch adults and the relationship between perception and production in this process. In Experiment 1, a four-day perceptual training protocol on the /æ/-/ɛ/ contrast was combined with related or unrelated production practice (participants said the target word or a phonologically unrelated word after each perceptual decision). There was improvement over the four days in perception and production, but no effect of type of production practice. Perceptual training can thus boost production learning even when participants have to produce the new vowels. Some individuals, however, are more successful in acquiring foreign sounds. In Experiment 2, Dutch students were followed longitudinally over their first year studying English at university. Preliminary results indicate that, in a passive oddball paradigm testing for a MisMatch Negativity (MMN) effect, students could discriminate English /æ/ from English /ɛ/ already at initial test, but Dutch /ɛ/ from English /ɛ/ only at final test. We will ask whether students who show a larger increase in MMN over time improve more in pronunciation of /æ/ and /ɛ/.

2:05

1pSCa4. Evidence for a 3-component model of speech production with phonology-extrinsic timing. Alice Turk (The Univ. of Edinburgh, PPLS, 3 Charles St., Edinburgh EH10 4ED, United Kingdom, a.turk@ed.ac.uk) and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

This talk will present evidence supporting a model of speech production which includes phonology-extrinsic timing, and consists of 1) a Phonological Planning Component to plan the goals for an utterance, including the segmental and prosodic structure for an utterance and non-grammatical goals such as speaking quickly or in a particular style, 2) a Phonetic Planning Component to plan the quantitative details of the acoustic goals and how they will be achieved articulatorily, and 3) a Motor-Sensory Implementation Component to ensure that the goals are achieved on time. Supporting evidence from the literature includes findings of greater timing precision at movement endpoints compared to other parts of movements, suggesting the separate control of the timing of movement endpoints compared to other parts of movement. This evidence presents a challenge to models in which all parts of a movement trajectory are controlled by the same equation of motion, but is consistent with models in which 1) abstract, symbolic phonological representations map onto spatial and temporal characteristics of movement endpoints, 2) movements are planned to reach the endpoints on time, and 3) speakers give priority to the accurate implementation of the part(s) of movement most closely related to the phonological goals.

2:25

1pSCa5. What the /f/? We're not done with fricatives yet. Integrating across time and frequency bands. Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, E11 SSH, Iowa City, IA 52242, bob-mcmurray@uiowa.edu), Marcus Galle (Psychol. Sci., Univ. of Texas Rio Grande Valley, Brownsville, TX), Ashley Farris-Trimble (Linguist, Simon Fraser Univ., Iowa City, IA, Iowa), and Michael Seedorff (Biostatistics, Univ. of Iowa, Iowa City, IA)

Fricatives represent an extreme version of lack of invariance. This illustrated by Jongman, Wayland and Wong's (2000) singularly comprehensive look at the acoustics of a set of speech sounds. They show that fricatives require dozens of cues; and each cue is affected by multiple context factors. Later work with this corpus solved these problems with simple models. But we're not done yet. I present new studies that raise two additional problems highlighted by fricatives. First, listeners must integrate cues over long timescales: frication cues may arrive several hundred milliseconds before the vocoid. I present several experiments that used eye-tracking to determine when cues are used. Unlike, prior results with stops and vowels, they show evidence for an encapsulated memory buffer preceding lexical access—lexical decisions are delayed until both cues arrive. Second, for fricatives, listeners must integrate information across distinct frequency bands. This was highlighted in a study of hybrid cochlear implant users who integrate low frequency residual acoustic hearing with high frequency electric hearing. Unexpectedly, hybrid listeners performed worse on fricatives than those with only electric hearing. Thus, the problem of combining information across frequency bands may be suboptimally solved in acoustic + electric CI users, suggesting new principles of cue-integration.

2:45

1pSCa6. Coupling tonetics and perceptual attunement: The psychophysics of lexical tone contrast salience. Denis K. Burnham (MARCS Inst. for Brain, Behaviour and Development, Western Sydney Univ., Locked Bag 1797, Penrith South, Sydney, NSW 1797, Australia, denis.burnham@westernsydney.edu.au) and Leher Singh (Psych., National Univ. of Singapore, Singapore, Singapore, Singapore)

As with the coupling phonetics and psycholinguistics by such luminaries as Sereno and Jongman, we couple (a) discrepant cross-lab results in infant lexical tone perceptual attunement studies and (b) an adult cross-language lexical tone perception study. Three trajectories of lexical tone perception over infants' first year have been found over labs, in which infants from differing language backgrounds have been tested with differing stimulus materials, viz., (i) Incremental (Singh's lab—Singaporean Mandarin vs Singaporean English language background infants, Singaporean Mandarin tone contrasts; Tsao's lab-Taiwanese and English language infants, Taiwanese Mandarin contrasts); (ii) Decremental (Burnham's lab—Australian English but not for Cantonese or Mandarin language infants, Thai contrasts; Yeung/Werker's lab—American English but not for Mandarin and Cantonese language infants, Cantonese contrasts); and (iii) U-shaped (Kager's lab—Dutch infants, Mainland Mandarin contrasts). To investigate these discrepancies, we tested discrimination of Singaporean Mandarin, Beijing Mandarin, Hong Kong Cantonese, and Bangkok Thai tone contrasts by Singaporean Mandarin, Beijing Mandarin, Hong Kong Cantonese, Bangkok Thai, and Sydney Australian English listeners. Despite some discrepancies, this coupling revealed some congruences between psychophysical level of difficulty of tone contrasts within and between languages in the adult study, and the different developmental trajectories in the infant studies.

Session 1pSCb

Speech Communication, Psychological and Physiological Acoustics: Coupling Phonetics and Psycholinguistics II (Poster Session)

Ann Bradlow, Cochair
Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL

Ratree Wayland, Cochair
Linguistics, University of Florida, 2801 SW 81st Street, Gainesville, FL 32608

Yue Wang, Cochair
Linguistics, Simon Fraser University, 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada

All posters will be on display and all authors will be at their posters from 3:30 p.m. to 5:30 p.m.

Contributed Papers

3:30

1pSCb1. Variability in speaking rate of native and non-native speakers. Melissa M. Baese-Berk, Kayla Walker (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu), and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

The bulk of the work on non-native speech has focused on average differences between L1 and L2 speakers. However there is growing evidence that variability also plays an important role in distinguishing L1 from L2 speech. While some studies have demonstrated greater variability for non-native than native speech (e.g., Baese-Berk & Morrill, 2015; Wade *et al*., 2007), others have demonstrated that under some circumstances non-native speech maybe *less*variable and that variability in non-native speech may shift as a function of many factors, including task (Baese-Berk & Morrill, to appear; Baese-Berk, Morrill, & Bradlow, 2016) and L1-L2 pairing (Vaughn, Baese-Berk, & Idemaru, to appear). In this study, we ask how variability manifests in L1 and L2 speech by speakers from a variety of language backgrounds. Specifically, we ask whether a speaker whose L1 speaking rate is highly variable is also highly variable in their L2. We also ask whether variability in speaking rate in L1 or L2 differs as a function of task (e.g., read vs. spontaneous speech) and complexity of the task (e.g., more or less complicated reading passages). The results of this study will inform our understanding of the myriad complex factors that influence non-native speech.

1pSCb2. Does native language temporal experience transfer to audio-visual synchrony perception? Dawn Behne (NTNU, Psych., Trondheim NO-7491, Norway, dawn.behne@svt.ntnu.no) and Yue Wang (SFU, Burnaby, BC, Canada)

The temporal alignment of what we hear and see is fundamental for the cognitive organization of information from our environment. Research indicates that a perceiver's experience influences sensitivity to audio-visual (AV) synchrony. We theorize that experience that enhances sensitivity to speech sound distinctions in the temporal domain would enhance sensitivity in AV synchrony perception. With this basis, a perceiver whose native language (L1) involves duration-based phonemic distinctions would be expected to be more sensitive to AV synchrony in speech than for an L1 which has less use of temporal cues. In the current study, simultaneity judgment data for the syllable /ba/ were collected with 23 steps of AV alignments: from audio preceding the video (audio-lead) to the audio and video being physically aligned (synchronous) to video preceding the audio (video-lead). Two groups of participants differing in L1 experience with phonemic duration were included: native speakers of Norwegian (binary phonemic quantity distinction) and English (no phonemic quantity distinction). Preliminary results based on measures the audio-lead threshold (ALT) support the hypothesis that native language experience may influence broad mechanisms of timing, such as those moderating AV synchrony perception. Findings contribute to understanding the underpinnings of experience and AV synchrony perception.

Invited Paper

1pSCb3. Building a multilingual ultrasound corpus. Kelly Berkson, Kenneth de Jong, Steven M. Lulich, and Malgorzata E. Cavar (Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

This paper presents a corpus of three dimensional ultrasound data focused on tongue shape during speech sound articulation. Our ultimate goal is to collect data pertinent to phonetic structures from as many languages as possible (at present upwards of 20 languages are represented) and to curate these data in an open access corpus that is freely available for use by other researchers. In this presentation, we review the structure of the corpus and present a series of case studies illustrating the ways in which three-dimensional data are being used to address questions of phonetic interest. Examples of such areas include: examining the articulation of laterals; determining the point of articulation for dorsal consonants; analyzing consonant and vowel coarticulation patterns; and elucidating how coupling of the tongue body with the tongue blade and root affect place and manner of articulation. Like the work of Allard Jongman and Joan Sereno, this project is inherently collaborative and includes ample opportunity for using guided investigation of targeted research questions to ease novice scholars into the research process.

1pSCb4. The role of social expectation in the perception of gay speech. Dominique A. Bouavichith (Linguist, Univ. of Michigan, Lorch Hall #455C, 611 Tappan St., Ann Arbor, MI 48109, dbouavichith@gmail.com)

Previous sociophonetic studies have characterized lengthened /s/ as one of several acoustic correlates of gay(-sounding) speech (Linville 1998; Rogers *et al.* 2000). It has also been shown that listeners adjust their perceptual expectations when given social information about a speaker (Strand & Johnson 1996; Nidzielski 1999; McGowan 2015). There is evidence that /s/ duration serves as a cue in the perception of a speaker's sexual orientation, but only when paired with another acoustic cue (Levon 2007). Can a social cue serve this role instead? Given these findings, how might the timecourse of lexical activation be affected as listeners are given social information about a speaker? The present investigation addresses these questions using eye tracking in a visual world paradigm to examine the effect of social expectation on the timecourse of lexical activation. Participants (N = 22) heard /CVs/ and /CVsC/ words with digitally lengthened /s/, in two test conditions. First, participants had no social information about the speaker; second, the speaker's sexuality was given implicitly. Gaze patterns differed based on listeners' experience levels with gay speech, with a higher latency to response for high-experience listeners. This suggests that social expectation varies as a function of listener experience with a sociolinguistic variety.

1pSCb5. Retention of speech sound production and perception in young international adoptees. Mirjam Broersma (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, P.O. Box 9103, Nijmegen 6500 HD, Netherlands, m.broersma@let.ru.nl), Wencui Zhou (Max Planck Inst. for PsychoLinguist, Nijmegen, Netherlands), and Anne Cutler (Western Sydney Univ., Sydney, NSW, Australia)

While international adoptees commonly report not remembering their birth language, studies have shown that with re-exposure they learn to perceive birth-language sounds faster (Choi *et al.*, PNAS, 2017; Pierce *et al.*, PNAS, 2014, Singh *et al.*, DevSci, 2011) and to pronounce them more accurately (Choi *et al.*, RSOS, 2017) than non-adopted controls. We assessed birth-language memories in much younger adoptees than tested before, investigating whether memories were episodic or abstract in nature and which process—imitation or perception—survived longest. Participants were (Experiment 1) 21 Cantonese and (Experiment 2) 25 Mandarin adoptees in the Netherlands and 47 Dutch control children, aged 4-11. They were trained on perception of Cantonese/Mandarin affricate and tone contrasts, and tested on perception and production (imitation; recordings assessed by native-speaker identification and rating). For perception, adoptees initially performed similar to controls but outperformed them after training, similar to findings for adoptees tested as adults. For production, however, adoptees already outperformed controls from the pre-test, in contrast to previous results for adult adoptees. Thus, while international adoptees retain abstract memories of their birth language phonology which helps them in relearning the sounds later in life, imitation ability stands the test of time better than perception ability.

1pSCb6. English listeners categorize murmured stops based on aspiration, not prevoicing. Luca Cavasso and Henny Yeung (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, lcavasso@sfu.ca)

Previous perceptual studies of English stop voicing focus on Voice-Onset Time (VOT). Aspiration is generally subsumed into VOT, yet [1] complicates this, evincing a trading relation between intensity of aspiration noise and VOT. Our study is the first to examine the role of VOT and aspiration in English listeners' perception of non-native plain and murmured stops. We recorded Marathi talkers producing /CVsV/ nonce words beginning with /t/, /t^h/, /d/, /d^h/, e.g. /d^{h}aːsaː/, or their velar counterparts. The following acoustic measures were taken for each token: — Duration of prevoicing — After Closure Time (ACT) [2], i.e., the interval between release and periodicity — Pre-Vocalic Interval (PVI) [3], which includes ACT and the murmured portion of the vowel — Intensity of aspiration noise Canadian English listeners rated the stops on a 6-point scale from voiced to voiceless. Only results for murmured stops are discussed here. Despite variability of prevoicing duration in these tokens (range = 200ms, sd = 43), the factor did not correlate significantly with a token's average rating (p>0.60). However, ACT (r=.53, p<0.001), PVI (r=0.39, p<0.001), and the mean intensity of aspiration noise (r = 0.58, p<0.02) did. Thus, aspiration, not prevoicing, best accounts for perceptual differences between murmured stops. [1] Repp, "Relative amplitude of aspiration noise…," *Lang. Speech* 22, 1979; [2] Mikuteit & Reetz, "Caught in the ACT…," *Lang. Speech* 50, 2007; [3] Berkson, "Capturing breathy voice…," *Kans. Work. Pap. Ling.* 33, 2012.

Invited Paper

1pSCb7. Linguistic experience and musical training in shaping Mandarin tone perception by trilingual non-native Cantonese listeners. Si Chen (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong 00000, Hong Kong, qinxi3@gmail.com), Yike Yang (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), Ratree Wayland, and Yiqing Zhu (Univ. of Florida, Gainesville, FL)

Mandarin tones are perceived categorically by native listeners, but not by non-native listeners (e.g., Francis *et al.*, 2003; Hallé *et al.*, 2004; Xu *et al.*, 2006). Vowel quality, stimulus duration, and language background also significantly contributed to categorical perception of tones among native and non-native listeners (Chen *et al.*, 2017). In comparison to pitch production, it was found that a relative shorter duration is required to perceive than to produce pitch contours, with non-tonal listeners needing longer duration to detect a change in the pitch direction. Duration asserts a stronger effect on between- and within-category discrimination patterns among tonal listeners. Fewer studies investigated the effects of stimulus duration and vowel quality in trilingual non-native speakers with and without musical training. Our study examines categorical perception of resynthesized pitch stimuli by 13 trilingual Cantonese musicians and 13 Cantonese non-musicians. We manipulated tones on both low and high vowels ([a] and [i]) to create 7-step, level-to-falling and level-to-rising pitch continua on both [a] and [i] vowels with 9 different duration values. Cantonese speakers participated in identification and same-different tasks.

Contributed Papers

1pSCb8. Investigating the representation of tonal alternations in context. Yu-Fu Chien (Chinese Lang. and Lit., Fudan Univ., Rm.701, West Wing Guanghua Bldg., Nm.220, Handa Rd. Yangpu District, Shanghai, Shanghai 200433, China, chien_yc@fudan.edu.cn), Hanbo Yan (Chinese Studies and Exchange, Shanghai Int.. Studies Univ., Shanghai, Shanghai, China), and Joan A. Sereno (Linguist, The Univ. of Kansas, Lawrence, KS)

Phonological alternations pose challenges to models of spoken word recognition in how surface information is mapped onto stored representations in the lexicon. In Mandarin, a full Tone3 (213) is reduced to an abridged tone (21) (half-third sandhi) when followed by Tone1, Tone2, or Tone4. In addition, Mandarin Tone3 is replaced by Tone2 when followed by another Tone3 (third-tone sandhi). Two experiments used auditory-auditory priming lexical decision to investigate the alternating representations of Tone3. In Experiment 1, targets consisted of disyllabic words with a half-third or full-third FIRST syllable. These targets were preceded by either a half-third prime, a full-third prime, or a control Tone1 prime. RT data showed facilitation effects for both half-third and full-third prime conditions, with no first syllable by prime type interaction. In Experiment 2, third-tone sandhi targets consisted of disyllabic words with a half-third or full-third SECOND syllable. These targets were preceded by either a half-third prime, a full-third prime, or a control Tone1 prime. The data also showed both half-third and full-third priming effects, without any second syllable by prime type interaction. The results suggest that Mandarin Tone3 is stored as half-third (21) and full-third (213) forms, with both of these tone3 phonological alternations activated.

1pSCb9. Do low intelligibility voices induce false memories? Stephanie Chung, Sophie Bishop, and Molly E. Babel (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, swychung@alumni.ubc.ca)

Spoken language is highly variable with phonetic variation that is attributable to multiple dimensions, including, but not limited to, differences in dialect, mood, native language, sociolinguistic dimensions, and talker-specific idiosyncrasies. Phonetic variation has documented effects on other aspects of linguistic processing. For example, Sumner and Kataoka found that phonetic variation affected semantic processing in semantic priming and false memory tasks, to which they concluding that socially dispreferred (i.e., voices or accents that are socially stigmatized or less desirable) phonetic variation received less attention in encoding, leading to a lack of priming and higher false memory rates [Sumner & Kataoka, 2013, JASA-EL, 134(6)]. In this study we compare the intelligibility and propensity to induce false memories in a between-subjects design. Intelligibility is assessed as response latency in a speeded shadowing task, and false memories are measured using standardized false memory lists where reporting of critical lure items not in the auditorily presented list are counted as false memories. The speaker set consists of five speakers of English, representing two local accents, two nonnative accents, and one native nonlocal accent. The results of this study contribute to our understanding of how phonetic variation affects word recognition and semantic processing.

1pSCb10. Acoustic-phonetic consequences of clear speech in Cantonese vowels and lexical tones. Angela Cooper (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5G4K2, Canada, angela.cooper@utoronto.ca), Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL), and Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

In adverse communicative contexts, speakers will modify their speech style to enhance intelligibility, which can involve a range of acoustic and articulatory adjustments. Prior work has focused on the influence of speaking style on spectral (e.g., vowel space expansion) and temporal distinctions (e.g., voice-onset time). Thus, this study investigated the extent to which such adjustments extend to the pitch dimension, namely lexical tone. Ten native Cantonese speakers were recorded producing five monosyllables with six Cantonese tones embedded in sentence contexts in both clear and conversational speaking styles. Acoustic analyses revealed that speakers produced two distinct styles, with longer vowel durations and greater vowel dispersion in clear relative to conversational speech. However, with regards to lexical tone production, the results indicated only a marginal re-adjustment of tones within the tone space as a result of clear speech. Specific contour tones, such as the low- and high-rising tones, saw increases in their F0 offset. However, mean F0, F0 range and tonal dispersion did not differ substantially between styles. These findings indicate that speakers maintain tonal stability across speaking styles, suggesting that, relative to segments, lexical tones may not be prioritized as phonological contrasts vital to enhance the overall intelligibility of the utterance.

1pSCb11. Speech acoustics, phonetics, and phonology unified by an optimal dynamic model of sound communication. Pierre Divenyi (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, pdivenyi@ccrma.stanford.edu), Mohamad Mrayati (Ctr. for Strategic Development, Riyadh, Saudi Arabia), and René Carré (none, Grenoble, France)

A book by us (Speech: A dynamic Process, de Gruyter, 2017) presents an optimal communication model for the production of sounds through dynamic deformation of a simple tube using the criterion that a maximum contrast between consecutive sounds must be achieved by expending a minimum effort. Its output corresponds well to fundamental articulatory speech data (places of articulation, F1-F2-F3 trajectories). When driven by a periodic source, the model generates natural-sounding vowels and, through constricting the tube at distinctive regions, also stop consonants. From this model's viewpoint, phonetic systems of vowels and consonants arise as the direct consequence of production dynamics capable of producing essentially any vowel or stop consonant. The good correspondence between characteristics of the model and those of speech leads us to conclude that (1) the model is explanatory, (2) because it is acoustically optimal so must be speech production, and (3) an acoustically anchored phonology will derive from it. Based on acoustics, the model thus identifies a natural link between phonetics, and phonology. Propositions of our approach have also been tested in perceptual studies.

Invited Paper

1pSCb12. The effect of minimal neighbors on acoustic detail in the production or real and nonce words in Italian. Olga Dmitrieva (Purdue Univ., 640 Oval Dr., Stanley Coulter 166, West Lafayette, IN 47907, odmitrie@purdue.edu) and Chiara Celata (Scuola Normale Superiore di Pisa, Pisa, Italy)

Previous research on English demonstrated that the existence of minimal neighbors in the lexicon can lead to the enhancement of the differentiating acoustic property, such as Voice Onset Time (VOT) in voiceless stops (Baese-Berk & Goldrick, 2009). In this study, the effect of minimal lexical neighbors (e.g., *cara* "dear" is a minimal neighbor for *gara* "match") on the acoustic realization of voicing and gemination was examined in real and nonce words in Italian. Thirty native speakers of Tuscan Italian read one of the two stimuli lists, one containing only geminated and voiced stimuli and the other containing only singleton and voiceless stimuli, such that a given participant was never exposed to both members of a minimal pair. The results showed that real words with minimal neighbors had longer negative VOT if voiced or positive VOT if voiceless than real words without minimal neighbors. However, the existence of a real minimal neighbor did not affect VOT duration in nonce words. Analysis of nasal and voiceless stop geminates is ongoing.

Contributed Papers

1pSCb13. Computer-vision analysis shows different facial movements for the production of different Mandarin tones. Saurabh Garg (Pacific Parkinson's Res. Ctr., Univ. of Br. Columbia, 442 East, 55th Ave., Vancouver, BC V5X1N4, Canada, srbh.garg@gmail.com), Lisa Tang, Ghassan Hamarneh (School of Comput. Sci., Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman (Dept. of Linguist, Univ. of Kansas, Lawrence, KS), Joan A. Sereno (Dept. of Linguist, Univ. of Kansas, Kansas City, KS), and Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

We aim to identify visual cues resulting from facial movements made during Mandarin tone production and examine how they are associated with each of the four tones. We use signal processing and computer vision techniques to analyze audio-video recordings of 21 native Mandarin speakers uttering the vowel /ɜ/ with each tone. Four facial interest points were automatically detected and tracked in the video frames: medial point of left-eyebrow, nose tip (proxy for head movement), and midpoints of the upper and lower lips. Spatiotemporal features were extracted from the positional profiles of each tracked point. These features included distance, velocity, and acceleration of local facial movements with respect to the resting face of each speaker. Analysis of variance and feature importance analysis based on random decision forest were performed to examine the significance of each feature for representing each tone and how well these features can individually and collectively characterize each tone. Preliminary results suggest alignments between articulatory movements and pitch trajectories, with downward or upward head and eyebrow movements following the dipping and rising tone trajectories, faster lip-closing toward the end of falling tone production, and minimal movements for the level tone.

1pSCb14. Perception of voiceless nasals in Mizo and Angami. Pamir Gogoi and Ratree Wayland (Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, pgogoi@ufl.edu)

This study is a perceptual analysis of place of articulation of voiceless nasals and their voiced counterparts in Mizo and Angami, two Tibeto-Burman languages spoken in North-East India. The voiceless nasals in Mizo and Angami differ in terms of their characteristics of voicing. In Mizo, there is a portion of voiced nasal murmur before the vowel starts, which has been compared to the voiceless nasals in Burmese (Bhaskararao & Ladefoged,1991). However, in Angami, the nasal murmur remains voiceless throughout (Bhaskararao & Ladefoged,1991; Blankenship *et al.*,1993). A previously performed acoustic analysis of cues for place of articulation in both the languages shows that the cues were more robust in the transition portion of the vowel following the voiceless nasal (Gogoi, 2018). The perception of voiceless nasals by native speakers have not yet been studied. However, existing literature on Burmese voiceless nasals have shown that there are sufficient acoustic cues in the voiced murmur portion of the voiceless nasals for discrimination of the place of articulation (Dantsuji,1986). This study investigates whether the perception results correspond to the acoustic analysis previously observed.

1pSCb15. The role of gender, ethnicity, and rurality in Mississippi back-vowel fronting. Wendy Herd, Joy Cariño, Meredith Hilliard, Emily Coggins, and Jessica Sherman (MS State Univ., 2004 Lee Hall, Drawer E, MS State, MS 39762, wherd@english.msstate.edu)

While tense back-vowel fronting is primarily associated with young women throughout most of the United States, it has been documented in the speech of both men and women from a wide range of ages in the South (Fridland, 2001; Clopper, Pisoni, and de Jong, 2005). However, most of the work documenting back-vowel fronting has been limited to White speakers and has not explored social variation related to ethnicity with the exception of Fridland and Bartlett (2006) who found that both White and Black speakers in Memphis exhibited back-vowel fronting. The present study uses isolated productions of *beat, boot, book, boat,* and *bought* by 73 participants from Mississippi to explore the role of gender, ethnicity, and rurality in back-vowel fronting. Gender and ethnicity—but not rurality—proved significant, and a significant gender by ethnicity interaction was found. These results indicate that White speakers from Mississippi front tense back vowels more than Black speakers and that White women are currently leading back-vowel fronting in this region. These results are consistent with previous findings that men participate in back-vowel fronting in the South, but they contradict Fridland and colleagues' findings that men lead /u/-fronting and that Black speakers are participating back-vowel fronting.

1pSCb16. Bilingual word familiarity in Cantonese and English. Lauretta Cheng (Linguist, Univ. of Michigan, Ann Arbor, MI), Khia A. Johnson, and Molly E. Babel (Linguist, Univ. of Br. Columbia, 2613 West Mall, Totem Field Studios, Vancouver, BC V6T 1Z4, Canada, khia.johnson@alumni.ubc.ca)

Bilingual speakers are typically unbalanced in their vocabularies in each language, with each language's lexicon being representative of the experiences and domains in which the bilingual uses that language. This can create a challenge in creating word lists for speech experiments, as the frequency counts from publicly available corpora do not represent the appropriate vocabulary domains. In this paper, we report on a word familiarity rating task in Cantonese and English that was designed to pretest stimuli in both languages to confirm the items were matched in familiarity for use in subsequent speech tasks. Participants fell into four different groups: Cantonese-English bilinguals who grew up in a Cantonese-dominant location, Cantonese-English bilinguals who grew up in an English-dominant location, North American English speakers with no Chinese language experience, and international English speakers with no Chinese language experience. All listeners were presented with blocks of Cantonese words/nonwords and English words/nonwords and were asked to rate the familiarity of each item. We compare these ratings across participant groups and in relation to corpora for each language to assess how the listener groups perform. We discuss these results in the context of designing stimuli lists that are appropriate for diverse, multilingual populations.

1pSCb17. Characterizing the coordination of speech production and breathing. Jeffrey Kallay (Univ. of Oregon, Eugene, OH), Melissa Redford, and Ulrich Mayr (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu)

A robust, positive correlation between inhalation depth and subsequent utterance length in read speech has been interpreted as an anticipatory effect. Since anticipatory posturing of the respiratory system during phrase preparation has important theoretical implications, the present study was designed to rigorously test the effect. Healthy college-aged participants learned by rote a passage with equal numbers of short and long sentences, which were randomized to create a zero (lag-1) autocorrelation sequence that controlled for priming effects. Rote learning was used to control for visual cues to utterance length. Multiple repetitions of the passage produced by six speakers were acoustically segmented into pause-speech intervals for preliminary analyses, with pause intervals coded for breath intake or no intake. Analyses on data that retained the experimental structure indicated a strong effect of preceding utterance length on the presence/absence of breath intake, but no effect of subsequent utterance length. Analyses of

breath pause durations indicated only significant negative correlations with utterance length, and no relation between intake duration and utterance length. Together, the results suggest only an effect of physiological recovery on speech breathing under conditions that better approximate spontaneous production than in previous studies. [Work supported in part by NIH award number R01HD087452.]

1pSCb18. The effects of second language proficiency on speech production in noise. Saya Kawase (Faculty of Sci. and Eng., Waseda Univ., 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, skawase@aoni.waseda.jp)

Speech production is a dynamic process and talkers produce variance of speech in order to maximize intelligibility and minimize articulatory efforts (cf. hyper- and hypo-articulation model). Prior research has shown that native (L1) talkers modify their speech produced in noise, such as increasing intensity, and changing formant frequency (especially F1 for vowel production). However, it is still unknown to what extent L2 talkers accommodate the environmental need. For example, speaking in noise is less familiar for L2 talkers, especially English as foreign language (EFL) learners. In addition, EFL learners lack interaction in their L2 usage, i.e., their speech may not be successfully modified considering listeners' facilitation. The current study thus tested this with Japanese EFL learners (high and low L2 proficiency). The target stimuli were English tense/lax vowel production (/i-ɪ/ and /u-ʊ/), which is known that Japanese learners of English tend to assimilate the spectral differences and instead use the durational contrast. Our preliminary results showed that both groups of the learners produced spectral enhancement of F1. However, only the high proficiency group used the temporal modification on /i/, potentially due to their faster speaking rate. These results suggest that different modification strategies were observed based on their L2 proficiency.

1pSCb19. Implicit biases in monolingual and bilingual speech perception. Ethan Kutlu and Ratree Wayland (Dept. of Linguist, Univ. of Florida, P.O. Box 115454, Gainesville, FL 32611-5454, denkutlu@ufl.edu)

Accented speech shows a great deal of variation from the "native" norms that can lead to processing difficulty. Previous research on listeners' performance on accented speech shows that the more varieties of accented speech the listeners are exposed to, the better their processing is (Baese-Berk, Bradlow, & Wright, 2013). Since many bilingualism studies suggest that bilinguals have advantages in allocating different resources such as attention during language processing (Costa, Hernández, & Sebastián-Gallés, 2008), there is a necessity to investigate bilingual listeners' processing of accented speech. In this pilot experiment, we will measure intelligibility and perceived degrees of accentedness of Tamil and British English presented in noise among American English monolinguals, and American English and Spanish Bilinguals. We will also perform different cognitive tests as well as an Implicit Association Test to see if prior implicit biases towards South Asian individuals affect the listeners' intelligibility and accentedness rating scores. We hypothesize that bilinguals will be more accurate in the intelligibility task compared to monolinguals due to their exposure to more than one language. However, it remains to be seen whether this advantage will be equally realized in both varieties of English accented speech.

1pSCb20. Adaptation of native clusters with non-native phonetic patterns. Harim Kwon (George Mason Univ., Fairfax, VA) and Ioana Chitoran (Université Paris Diderot, 175 rue du Chevaleret, Paris 75013, France, ioana.chitoran@dartmouth.edu)

Non-native consonant clusters are modified to conform with native phonotactics. This study investigates how onset clusters that are phonotactically licit, but have non-native phonetic patterns, are adapted to match the native patterns. We tested Georgian speakers in a sentence completion task using CCV/CVCV sequences produced by a French talker. Georgian differs from French in having (1) longer inter-consonant timing lag for CCVs, and (2) default initial prominence for CVCVs. The long inter-consonantal lag often results in a transitional vowel in Georgian. Georgian participants (n = 11) first saw the target CCV/CVCV sequences embedded in a Georgian carrier phrase and read the phrase aloud (baseline). Then they heard the target sequences produced by a French talker while seeing the Georgian carrier phrase with an empty slot, and produced the phrase completed with the heard targets (test). Participants' test productions reflected modifications of French targets towards their native phonetic patterns, not only by producing occasional transitional vowels that are absent from French CCV targets, but by deleting the unstressed first vowel in French CVCVs (/pøta/ produced as /pta/). We claim that the effects of native language on adaptation of non-native sequences are not limited to their segmental composition, but also involve their phonetic implementation.

1pSCb21. The relation between production and perception of Mandarin tone. Keith K. W. Leung and Yue Wang (Linguist, Simon Fraser Univ., Robert C. Brown Hall Bldg., Rm. 9201, 8888 University Dr., Burnaby, BC V5A 1S6, Canada, kwl23@sfu.ca)

This pilot study explores the potential production-perception relation of Mandarin tones by examining the association between the acoustic features of native Mandarin tone production and perception of such features, using Tone 2 (rising tone) as a testing case. The perception stimuli were resynthesized by varying F0 onset (spectral feature) and F0 turning-point location (temporal feature) orthogonally forming a grid of tone contours. The endpoints of these dimensions were determined by natural Tone 1 and 3 productions, presumably corresponding to perceptual boundaries for Tone 2. A parabola was fitted to the resynthesized tones to determine the overall shape of the tone contour. Native Mandarin participants were asked to identify the best exemplar of Tone 2. The acoustic analysis of Tone 2 productions by the same participants included measurements of the same features. Preliminary correlation analysis showed a positive production-perception correlation for the temporal feature of F0 turning-point location, but no significant correlation for the spectral features of F0 onset and F0 contour shape, indicating F0 turning-point location as a critical perceptual cue for Tone 2. Further research will include additional tones and additional acoustic features found to characterize these tones to identify the nature of production-perception relation of Mandarin tones.

1pSCb22. Cue enhancement and effects in [l] and [n] identification in L2 English. Bin Li (Linguist and Translation, The City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon Tong 000, Hong Kong, binli2@cityu.edu.hk)

[l] and [n] in English sharing certain articulatory characteristics are acoustically similar (Ladefoged 2003, Lass 1996), which has been reported to cause confusion in consonant recognition (Mermelstein 1977, Espy-Wilson 1992, Pruthi 2004), and adds significantly to the perceptual challenge that Chinese L2 learners of English are faced with (Li 2006). Young Cantonese speakers, for example, often get confused with the English contrast because /l/ and /n/ are considered in free variation and in the process of merging into one in contemporary Cantonese. Despite the articulatory and acoustic similarities, [l] and [n] differ in many ways such as consonantal duration and changes at the constriction release (Li, Zhang, Wayland 2012). This study modified temporal and spectral correlates of the two consonants at various positions in a word, so as to test if cue enhancement could affect L2 consonant perception. University students in Hong Kong were recruited to identify sounds of the contrasting pair in English. Their results suggest that lengthening the consonant and consonant-vowel transition may help the learners better identify the sounds.

1pSCb23. Categorical perception of Mandarin tones by Cantonese trilingual children. Shiyue Li, Si Chen, and Angel Chan (CBS, The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 999077, Hong Kong, shiyue.li@connect.polyu.hk)

It is generally argued that children's tone perception gradually becomes adult-like after age 6. In addition, previous studies suggest that stimulus duration, vowel quality, and musical training significantly contribute to categorical perception of pitch directions in both tonal and non-tonal language (Chen *et al.*, 2017; Zhao & Kuhl, 2015). However, little is known about how musical training may shape categorical perception of trilingual children, and how stimulus duration and vowel quality come into play. In this paper, we investigate these effects on the categorical perception of Mandarin tones based on Cantonese trilingual children. Specifically, we created rising and falling Mandarin continua on high and low vowels with three duration values. Based on some pilot data, the vowel quality plays a vital role in the categorical perception of rising tones but not in the falling tone pitch directions. Children with professional musical training show much more categorical perception than children without musical training. Specifically, musically trained children show more between-category discrimination than within-category discrimination. Moreover, they are more sensitive to stimulus and are faster in terms of reaction time. Our findings will be compared with previous studies on categorical perception by adults and all significant effects will be discussed.

1pSCb24. Mandarin learners' ESL fricatives. Hua Lin and Junyu Wu (Dept. of Linguist., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, hualin@uvic.ca)

Although English interdentals have been much studied in L2 acquisition, other English fricatives have not received comparable attention. The assumption is that the other fricatives, e.g., [s] which often have L1 counterparts do not pose a problem for ESL learners. However, it is now well understood that similarity between L1 and L2 counterparts may be more problematic in perpetuating an L2 accent. To gain insights into the issue, Mandarin speakers' production of both ESL and native fricatives and their perception of the ESL fricatives is collected and analyzed. The results show that the lack of a problem is true for perception which shows ceiling performance. Production-wise, all ESL voiceless fricatives, not just [θ], are problematic for the Mandarin speakers, judged by a native speaker and measured acoustically on spectral moments and peak. Typically, these fricatives are not on target, but are produced more or less in the L1 fashion. A surprise finding is also made that an L1 counter part may not be the choice of a match; e.g., English [s] which has a close Mandarin counterpart is found replaced with [θ] or [ʃ] by some participants. The paper also discusses the production-perception relation over the findings on the fricatives.

1pSCb25. Perception and production of syllable-tone homophones by second language learners of Chinese. Jiang Liu (Lang., Literatures and Cultures, Univ. of South Carolina, 1620 College St., 917 Humanities Office Bldg., Columbia, SC 29208, jiangliu@mailbox.sc.edu) and Seth Wiener (Modern Languages, Carnegie Mellon Univ., Pittsburgh, PA)

This study examined how the relatively high degree of tonal homophony in spoken Mandarin affects learners' perception and production of new syllable-tone words. Seventeen English-speaking-learners of L2 Chinese participated in a 3-day word-learning task in which pictures were paired with spoken syllable-tone words. Stimuli included 32 monosyllabic Chinese words (e.g., yu3 "feather"). 16 words corresponded to previously learned syllable-tone homophones (e.g., yu3 "language"); 16 words were previously unlearned combinations (e.g., yu4 "jade"). Learning was assessed on day 3 using a 4-alternative-forced-choice identification task and a picture-naming task. Twelve Chinese-L1 speakers identified the learners' productions. Learners were overall more accurate at identifying words that had no previously learned homophone (94%) than words that lacked homophones (90%). The opposite trend was observed in production: learners were overall more accurate at producing words that corresponded to previously learned homophones (77%) than words that lacked homophones (72%). Mixed-effects logistic regression models revealed no statistical difference of the log odds of correct identification or production. These results indicate that despite encountering considerable tonal homophony in speech, learners did not gain advantage in perceiving or producing new words with homophones. The phonologically marked T2-T3 contrast was produced with the lowest accuracy relative to other tonal contrasts.

1pSCb26. The perception-production link in sibilant convergence. Yanyu Long (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, yl2535@cornell.edu)

Many theories suggest a link between perception and production, but evidence for such a link is inconsistent. This study addresses this issue by looking at the relationship between perception of sibilants before [u] or [a] and production change when imitating words starting with a lower-frequency [s] before [u] or [a]. Sibilants were measured by center of gravity. The results at first showed no relationship between production change and perception. However, a closer look revealed that some participants converged to the stimuli while others did not. Mixed effect models showed that for people who converged, larger difference between the perception distribution and the stimuli corresponds to more convergence. This is consistent with the observation that people converged more in the [sa] words, since [s] before [a] has higher perception distribution, hence more distant from the stimuli. No difference was observed between the [u] and [a] contexts in terms of the coefficients of perception in predicting convergence. The results show that no matter what the phonetic context is, inputs are recoded in terms of their distance from the perception distribution before influencing production. With larger such distance, the chosen production targets are farer away from the baseline, resulting in more convergence.

1pSCb27. Sub-articulatory interactions in palatalization processes. Malgorzata E. Cavar (Linguist, Indiana Univ., Bloomington, IN) and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Palatalization is a widespread phenomenon across the world's languages and is typically conceived articulatorily as a tongue body raising and/or fronting gesture that is spread from high and/or front vowels to neighboring consonants. Oddly, palatalization of labial consonants (which allow the greatest degree of gestural overlap with neighboring high and/or front vowels) is typologically least common, suggesting that palatalization processes do not primarily result from gestural overlap, but instead arise from the resolution of incompatible lingual "sub-articulations." In particular, articulatory data from Polish and Russian palatalized consonants point to a critical role for tongue root interactions with tongue blade and dorsum gestures. The products of these resolved sub-articulatory interactions can become lexicalized and reconfigured by later generations of language-learners in response to the interaction of phonological pressure to maintain contrasts and new palatalization processes. The etymological history of the Proto-Slavic stems **menk-* and **ment-* illustrates these interactions. The velar consonant in **menk-* was palatalized by a following high, front vowel to a "soft" posterior affricate. Subsequently, the coronal consonant in **ment-* was palatalized as well (Modern Polish *mącić*). To maintain the contrast between the sound categories, the palatalized velar was reconfigured as a post-alveolar "hard" consonant (*mączyć*).

1pSCb28. Gender, the individual, and intelligibility. Daniel McCloy (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu), Laura Panfili (Dept. of Linguist., Univ. of Washington, Seattle, WA), Cornelia John, Matthew Winn (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Richard Wright (Dept. of Linguist., Univ. of Washington, Seattle, WA)

In the clinic and in the laboratory, opinions differ on the relative intelligibility of the speech of women and men. However, the effect of gender alone has rarely been studied explicitly. Here we present a study of 30 talkers (15 male) and 32 listeners assessing intelligibility of a 180-sentence subset of the IEEE sentences presented in steady-state speech-shaped noise. Four signal-to-noise ratios (−4, −2, 0, +2 dB SNR) were tested with 45 sentences each. Results showed substantial overlap between intelligibility scores for each gender. Although standard statistical approaches show a slight advantage for female talkers at all SNRs, post-hoc analyses indicated that the gender effect is an artifact driven by a few particularly unintelligible males. These results do not address intrinsic gender-related differences in speech intensity, or in the ability to overcome background noise by speaking clearly, but suggest that gender-related differences are negligible when those factors are controlled. More generally, even with a large sample of talkers, the high degree of talker-intrinsic variability in intelligibility can lead to conclusions that do not generalize to the population of interest, an issue that could affect comparisons rooted in gender, dialect, or other social factors.

1pSCb29. Comparing acoustic and articulatory targets in short-term visual feedback training of non-native vowels. Sonya Mehta (The Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, naya@utdallas.edu) and William F. Katz (The Univ. of Texas at Dallas, TX)

Theories of speech production aim to explain how talkers express abstract linguistic forms as audible events that are intelligible to both speaker and listener. The relationship among planned units of speech, their articulatory implementation, and their acoustic consequences is thus a key issue in speech research. The work reported here is part of a larger project designed to investigate the effects of visual acoustic and visual articulatory feedback on second language (L2) learners' production and perception of non-native speech sounds. L2 talkers from a variety of language backgrounds practiced producing an English vowel, /æ/, while receiving visual feedback on either (1) first and second formant frequencies, provided by a real-time spectrographic display, or (2) tongue back position, shown using a talker-driven tongue avatar. Kinematic data were recorded using an electromagnetic articulograph (EMA) system that tracked tongue midline and lateral movement during vowel productions. Pronunciation accuracy was analyzed by calculating acoustic and kinematic Mahalanobis distances between L2 productions and target (native talker) exemplars. Initial analyses of a single subject's data showed that both types of visual feedback training improved pronunciation, suggesting that both acoustic and articulatory information are recruited during vowel production.

1pSCb30. Gender normalization in fricative perception: Generational differences. Benjamin Munson, Mara Logerquist, and Melanie Putman (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Listeners identify fricatives ambiguous between /s/ and /ʃ/ differently depending on whether they believe that the talker is a woman or a man (Strand & Johnson, 1996; Munson, 2011; Winn, Rhone, Chatterjee, & Idsardi, 2013). Recently, Munson and Logerquist (2017, J. Acoust. Soc. Am. 141, 3982) examined the influence of imputed talker gender on fricative perception across four experimental conditions in a relatively large group (n = 99) of young adults. Surprisingly, Munson and Logerquist were unable to replicate the original finding, despite using stimuli that resembled those from earlier studies very closely. In the current study, we test the hypothesis that Munson and Logerquist's failure to find an effect of imputed gender was because of generational changes in the perception of gender in speech. We do this by comparing the performance of younger, college-aged listeners in Munson and Logerquist to the performance of a new cohort of listeners aged 33–48. Data collection for the latter cohort is ongoing. A finding that these listeners show effects of gender on fricative identification similar to those in previous studies will support the idea that Munson and Logerquist's failure to replicate this finding is due to generational changes in the perception of gender through speech.

1pSCb31. Non-native but not native listeners rely on exemplars for comprehending reduced pronunciation variants. Annika Nijveld, Louis ten Bosch, and Mirjam Ernestus (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, PO Box 9103, Nijmegen 6500 HD, Netherlands, a.nijveld@let.ru.nl)

This study investigates the representations of words' pronunciations in the mental lexicon, and how they are used in spoken word recognition by native (L1) and non-native (L2) listeners. We report two long-term repetition priming lexical decision experiments, each conducted with native English listeners, Dutch learners, and Spanish learners of English. We tested whether the listeners recognized word repetitions more quickly and/or more accurately when the first ('prime') and second ('target') occurrence of the word shared surface details (e.g., the voice of the speaker) compared to when they did not. If so, this suggests that listeners retain word tokens with their acoustic details in their memories (in the form of an exemplar). We found that in one of the experiments, L2 listeners relied more on exemplar representations than L1 listeners did. Specifically, larger exemplar effects arose for the non-native listeners in the experiment in which we included reduced pronunciation variants resulting from acoustic schwa reduction in English (e.g., *b^{a}lloon*). Reduced pronunciation variants are highly difficult to process for non-native but not for native listeners. Our finding suggests that exemplars play a role in spoken word recognition under specific circumstances only, such as under challenging listening conditions for L2 learners.

1pSCb32. Effect of native Korean dialects on Chinese learners' perception of lexical tones. Zhen Qin (Shanghai Jiao Tong Univ., 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045, qinzhenquentin@yahoo.com)

Second-language learners (L2ers) weight cues (e.g., pitch) as a function of how the cues are used in the native language. This study investigates the effect of native dialect on the use of Chinese (i.e., Mandarin Chinese) tonal information by native speakers of Seoul Korean (SK) and Kyungsang Korean (KK). While SK does not use pitch to realize lexical prosody, KK uses pitch to realize lexically contrastive words. Intermediate-to-advanced SK-speaking or KK-speaking L2ers of Chinese (at the same Chinese proficiency) completed a forced-choice tone identification task and a speeded AX tone discrimination task. In the identification experiment, participants heard natural tonal stimuli carried by multiple syllables; in the discrimination experiment, the tonal stimuli were resynthesized to model on natural citation forms of two native speakers (one male, one female), and were superimposed on the vowel /y/. Data collection, with 15 SK listeners and 15 KK listeners to be tested, is ongoing in Shanghai. KK listeners are predicted to have a higher accuracy than SK listeners in identifying tones, and to weigh pitch contour more than SK listeners in MDS analyses of RTs when discriminating tones. If our hypothesis is correct, native dialect is suggested to be considered in L2 speech perception models.

1pSCb33. Productive pitfalls in the phonetic pursuit of psycholinguistic questions. Melissa Redford (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu), Jeffrey Kallay (Univ. of Oregon, Eugene, OH), and Jill Potratz (Univ. of Oregon, Eugene, OR)

Coarticulation in adult speech spans multiple segments, including across syllable and word boundaries. Little in the way of comparable data is available for child speech, where the focus has been on within-syllable anticipatory effects. Still, the developmental studies indicate that anticipatory coarticulation within a syllable is as strong in child speech as in adult speech, and perhaps stronger. This finding is consistent with the hypothesis that words are fundamental units of production under the long-held assumption that anticipatory coarticulation reflects speech motor planning. Our current work leverages this same assumption to investigate age-related changes in the structure of the speech plan. A related goal is to characterize the development of long-distance anticipatory coarticulation. For example, we are testing the hypothesis that younger children's plan consists of more fully individuated word-sized production units than the plan guiding older children's and adults' speech in sentence elicitation experiments, where the initial boundary of a production unit is operationalized as the onset of lip rounding. The work has propelled us to develop a novel psycholinguistic measure of coarticulation, and is now leading us to question the concept of a speech plan and the nature of anticipatory coarticulation. [Work supported by NIH award number R01HD087452.]

1pSCb34. Lexically dependent estimation of acoustic information in speech. Charles Redmon and Allard Jongman (Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm 427, Lawrence, KS 66046, redmon@ku.edu)

Current estimates of the relevant acoustic information in speech are typically inventory-based. They assume a set of phonemes or featural/gestural decompositions of phonemes exists in a language, and proceed to identify and rank/weight the set of acoustic parameters sufficient to distinguish a phonemic contrast based on data from instantiations of the sounds/categories comprising those contrasts in either a controlled set of real words in the language or in nonword syllables. Acknowledging that such contrasts ultimately derive from lexical oppositions, we present an alternative, complementary approach to the quantification of acoustic information in speech: one that is lexically based. Using acoustic data from the Massive Auditory Lexical Decision project (Tucker *et al*., 2018)—over 26,000 unique words produced by a single speaker of Western Canadian English—and listener identification data on a representative 240-word sample of that lexicon, we

define a weighted network of the phonological lexicon (cf. Vitevitch, 2008), where the weights on links between minimal pairs correspond to the predicted acoustic similarity from a model fit to the listener error distributions. From this network the distributed, "global" information contributed by individual parameters operating in an ensemble of lexical oppositions can be estimated from changes in network entropy under perturbations of those parameters.

1pSCb35. Effects of cochlear-implant simulation on processing of vowel sequences by young normal-hearing listeners. Catherine L. Rogers, Jenna Vallario (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu), and Gail Donaldson (Commun. Sci. and Disord., Appalachian State Univ., Tampa, FL)

To better understand the effects of listening environment on efficiency of phonetic processing, the present study examined the effects of signal degradation on phonetic processing of two-syllable sequences by normal-hearing listeners. Auditory temporal-order processing of American-English vowel sequences was compared across two listening conditions, each presented to a separate group of young, normal-hearing listeners: 1) unprocessed resynthesized stimuli and 2) stimuli that had been processed to simulate the signal produced by a 16-channel cochlear implant (CI). Using the methods of Fogerty, Humes and Kewley-Port [2010, *J. Acoust. Soc. Am.*, **127**, 2509-2520], 70-ms resynthesized versions of the syllables "pit, pet, put," and "pot" were presented in a two-syllable temporal-order processing task. Task difficulty was increased by decreasing syllable-onset asynchrony (SOA), i.e., the duration between syllable onsets. SOA thresholds for accuracy of syllable-sequence identification were estimated using the method of constant stimuli on each of four 72-trial blocks. Data analyzed to date show a threshold difference of approximately 20 ms between the unprocessed and CI-processed listener groups, or a difference in threshold of a factor of two or greater. Results will be discussed with regard to implications for phonetic processing of speech in challenging listening environments and practical implications for CI users.

1pSCb36. Effect of distributional shape on learning a target sound. Emily Sadlier-Brown and Carla L. Hudson Kam (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, e.sadlier-brown@alumni.ubc.ca)

Vowel pronunciations (measured in Hz) form distributions that vary from normally distributed to quite skewed. Labov (2001) noted that vowels undergoing change tend to be skewed, while stable ones are not. We ask if the different distributions are causally related to change. We exposed participants (n = 238) to a positively skewed, negatively skewed, or normal distribution of pure tones varying in pitch (to mimic how vowels vary in quality). Participants were told they were listening to notes played by amateur musicians who'd been aiming for the same note. In each trial, participants listened to 20 tones then played the note they thought was the target. Output pitches were compared to the input. The question was whether learners would play a tone corresponding to the mean or the mode of the set or would instead "shift" the note. In the two conditions analyzed so far (normal and positively skewed), participants output pitches that were slightly higher than the mean of the input set, indicating shift in both conditions; however, the difference between conditions is not significant. There is also an effect of the final note in the set, evoking the well-established recency effect in memory. Analysis of the third condition is ongoing.

1pSCb37. Realtime integration of acoustic input and semantic expectations in speech processing: evidence from electroencephalography. McCall E. Sarrett (Psychol. and Brain Sci., Univ. of Iowa, Spence Labs. of Psych., 308 Iowa Ave., Iowa City, IA 52240, mccall-sarrett@uiowa.edu), Efthymia Kapnoula (Basque Ctr. on Cognition, Brain, and Lang., San Sebastian, Spain), and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

A critical question in speech perception is the relative independence of perceptual and semantic processing. Answering this requires addressing two issues. First, does perceptual processing complete before semantic processing (discrete stages vs. continuous cascades)? Second, do semantic expectations affect perceptual processing (feedback)? These questions are difficult to address as there are few measures of early perceptual processing for speech. We extend a recent electroencephalography (EEG) paradigm which has shown sensitivity to pre-categorical encoding of Voice-Onset Time (VOT; Toscano *et al*., 2010). By measuring the timecourse over which perceptual and semantic factors affect the neural signal, we quantify how these processes interact. Participants (N = 31) heard sentences (*Good dogs also sometimes—*) which biased them to expect a target word (*bark* rather than *park*). We manipulated VOT of the target word and coarticulation leading to it. A component-independent analysis determined when each cue affects the continuous EEG signal every 2 msec. This revealed an early window (125–225 msec) sensitive exclusively to bottom-up information, a late window (400–575 msec) sensitive to semantic information, and a critical intermediate window (225–350 msec) during which VOT and coarticulation are processed simultaneously with semantic expectations. This suggests continuous cascades and early interactions between perceptual and semantic processes.

1pSCb38. Developing new pre-lexical processing skills in adults. Anna M. Schmidt (Speech Path. & Aud., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

Studies of adults learning to decode words may provide a window on understanding of the pre-lexical processing of words as learning progresses. Second language (L2) learners (for example) have been found to assimilate new phones to first language (L1) phonemes when asked for the closest sound in their languages (e.g., Schmidt, 1996). Ratings of fit with L1 phonemes was related to details of acoustics of stimuli, L1 acoustics, and vowel context suggesting that L1 phonetic details rather than phoneme categories were compared. What happens after attention has been focused on acoustic/articulatory characteristics of these new sounds (as in training studies) and learning progresses? Do adults establish new phoneme categories to aid in word processing or is it more likely that native categories are expanded based upon developing within L1 category perceptual skills and orthographic word prediction skills. Evidence will be reviewed indicating that learned production of L2 sounds can be become native-like but L2 perception outcomes remain non-native like suggesting that new phoneme categories are not formed. Thus, new learned words can be accurately produced but success in pre-lexical processing of new heard words will depend upon linguistic factors such as context.

1pSCb39. Recalibration dominated by the right ear. Mark Scott (Dept. of English Lit. and Linguist, Qatar Univ., Doha 27419, Qatar, mark.a.j.scott@gmail.com)

Recalibration is a phonetic learning process by which the perceptual boundaries between speech sounds are adjusted. In a typical recalibration experiment, ambiguous sounds are paired with videos that disambiguate the ambiguous sounds to one category or another, the perceiver learns the association and, when the video is absent, continues to perceive the ambiguous sounds as the learned categories. The current experiment tests whether recalibration is stronger in the right or left ear. The right ear has a direct link to the left hemisphere, where such perceptual learning likely occurs, and so we might expect the right ear to be *more* susceptible to recalibration. However, the right ear is less influenced by visual information so we might expect it to be *less* susceptible to such visual-based recalibration. In this experiment, participants heard an ambiguous speech sound (between /b/ and /d/) which was rendered unambiguous by accompanying video of a face pronouncing one of these sounds. In each block, the audio was presented exclusively to the left or right ear. After multiple exposures, participants' perception of the ambiguous sound (without accompanying video) was tested. Data collection is ongoing, but currently there is a strong trend towards more recalibration in the right ear.

1pSCb40. Linking production and perception of clear speech. Joan A. Sereno, Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS), Yue Wang (Linguist, Simon Fraser Univ., 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada, yuew@sfu.ca), Ghassan Hamarneh, Lisa Tang (Comput. Sci., Simon Fraser Univ., Vancouver, BC, Canada), Saurabh Garg (Univ. of Br. Columbia, Vancouver, BC, Canada), Paul Tupper (Mathematics, Simon Fraser Univ., Burnaby, BC, Canada), Bob McMurray (Psych., Univ. of Iowa, Iowa City, IA), Charles Redmon, Yuyu Zeng (Linguist, Univ. of Kansas, Lawrence, KS), Beverly Hannah, Keith K. W. Leung, and Sylvia Cho (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Speech communication can adopt different styles as a function of speaking environments and communicative needs. In auditorily or visually challenging contexts, speakers often alter their speech production using a clarified, hyper-articulated speech style with the intention of enhancing speech intelligibility. Such modifications may result in perceptible articulatory and acoustic changes. Questions thus arise as to whether and what clear-speech modifications facilitate perception. This presentation surveys recent research conducted in our labs, investigating clear-speech production and its associated effects on perception. In a series of three-stream studies, this research relates analyses of visible articulatory features using computer image-processing techniques, measurements of acoustic properties, and perceptual patterns of clear-speech segments and suprasegmentals by native and non-native perceivers. Results reveal that clear (relative to plain) speech modulates different and compensatory articulatory-acoustic cues within each sound category to enhance intelligibility. However, our results also show that clear-speech modifications that reduce phonemic category distinctiveness inhibit intelligibility. These findings indicate that clear-speech effects are governed by the collateral principles of within-category cue enhancement and maintenance of category distinctiveness.

1pSCb41. Weak adaptation to foreign-accented voice-onset-time distribution. Tifani Biro (Penn State Univ., University Park, PA), Seulgi Shin, Yuyu Zeng, and Annie Tremblay (Linguist, Univ. of Kansas, 541 Lilac Ln. Blake Hall, Rm. 427, Lawrence, KS 60045-3129, atrembla.ku@gmail.com)

Listeners quickly adapt to foreign-accented speech in transcriptions and word/sentence judgments (Bradlow & Bent, 2008; Clarke & Garrett, 2004; Sidaras, Alexander, & Nygaard, 2009). However, continuous spoken-word recognition measures such as eye tracking have revealed limitations on foreign-accent adaptation. For example, Trude, Tremblay, and Brown-Schmidt (2013) found that English listeners' adaptation to a second-order constraint in a Quebec French talker's accent (/i/ = [ɪ] before coda consonants except voiced fricatives) was limited. This study investigates whether foreign-accent adaptation is less limited when the mapping between the accented and underlying words does not require higher-level inferencing. English listeners completed an eye-tracking experiment in which they heard a (female) French talker and a (male) English talker. Target and competitor words began with a voiced or voiceless stop and were otherwise temporarily ambiguous (e.g., *timber*and *dimple*). All stops were resynthesized: The French talker's stops were prevoiced (voiced) and short-lag (voiceless) and the English talker's stops were short-lag (voiced) and long-lag (voiceless). Preliminary results suggest an effect of voicing only for the French talker, with voiceless-stop targets generating more competition than voiced-stop targets. Importantly, this effect decreased only slightly from the first to the second half of the experiment, suggesting weak foreign-accent adaptation.

1pSCb42. Perception of word boundaries in spontaneous speech. Jiaer Tao, Francisco Torreira, and Meghan Clayards (McGill Univ., 1085 Ave Dr. Penfield, Montreal, QC H3A 1A7, Canada, meghan.clayards@mcgill.ca)

Studies of lab speech have identified several acoustic variables that listeners use to identify word boundaries (e.g. allophonic variation, segment duration), however spontaneous speech may not contain the same acoustic signals. Using data from a recent study we tested whether disambiguating information is also available to listeners under more typical conditions. 15 pairs of target phrases only differing in the placement of word boundaries, e.g., *grey#day vs. grade#A* were elicited under read (146 tokens) and spontaneous speech conditions (316 tokens) from multiple talkers who were not aware of the ambiguities. Phrases were played in isolation to 30 Native English listeners in a 2AFC segmentation task (e.g. *grey day* vs. *grade A*). Accuracy was above chance for both read (80.9%) and spontaneous speech (73.1%). Accuracy varied according to the critical consonant type and was highest for voiceless stops followed by nasals, voiced stops, fricatives and clusters. Only two items (*pierced ears* vs. *peer steers* and *beef eater* vs. *bee feeder*) were at chance for spontaneous speech. Measures of word-initial lengthening were good predictors of performance but listeners outperformed predictions from these measures indicating they weren't the only cues.

1pSCb43. Use of tonal information in French and English listeners' segmentation of Korean speech. Annie Tremblay, Seulgi Shin (Linguist, Univ. of Kansas, 541 Lilac Ln. Blake Hall, Rm. 427, Lawrence, KS 60045-3129, atrembla.ku@gmail.com), Sahyang Kim (English Education, Hongik Univ., Seoul, South Korea), and Taehong Cho (English Lang. & Lit., Hanyang Univ., Seoul, South Korea)

(Seoul) Korean has been analyzed as having an Accentual Phrase (AP) with a L(HL)H tonal pattern for APs beginning with a lenis segment (Jun, 1998). This tonal information modulates Korean listeners' speech segmentation (Kim & Cho, 2009; Tremblay, Shin, Kim, & Cho, 2018). This study investigates whether French- and English-speaking late learners of Korean can also use this tonal information. French has similarly been analyzed as having an AP with a L(HL)H tonal pattern (Jun & Fougeron, 2002). In contrast, English words are often stressed initially (Cutler & Carter, 1987), with nuclear-pitch-accented words beginning with a H tone (Beckman, 1986). French, English, and Korean listeners completed an eye-tracking experiment. They heard sentences containing a temporary ambiguity between an AP-initial target (*saesinbu-ga]*$_{AP}$*[masul-eul* 'the-new-bride-subj magic-obj') and a disyllabic competitor spanning the AP boundary (*gama*"palanquin"). Stimuli were resynthesized to orthogonally manipulate the AP-final and AP-initial tones (H/L). Results show that learners eventually used the AP-final and AP-initial tones like Korean listeners, but French listeners made earlier use of both tones than English listeners, even if the AP-initial tone is less crucial to French speech segmentation (Tremblay, Broersma, Coughlin, & Choi, 2016). Speech segmentation is thus very adaptive, even among late learners.

1pSCb44. Identifying the distinctive acoustic cues of Mandarin tones. Paul Tupper (Dept. of Mathematics, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, paul_tupper@math.sfu.ca), Keith K. W. Leung, Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman, and Joan A. Sereno (Univ. of Kansas, Lawrence, KS)

Using mathematical modeling, this study aims to characterize distinctive acoustic features of Mandarin tones based on a corpus of 1013 monosyllabic words produced by 21 native Mandarin speakers. For each tone, 22 acoustic cues were extracted. Besides standard F0, duration, and intensity measures, further cues were determined by fitting two mathematical models to the pitch contours. The first is a broken-line model, which models the contour as a continuous curve consisting of two lines with a single breakpoint. The second model is a parabola, which gives three parameters: a mean F0, an F0 slope, and an F0 curvature. Using Cohen's d, we identify which of the 22 cues are important for distinguishing each tone from the others for all speakers, as well as identifying cues that are used idiosyncratically by particular speakers. Although the specific cues that best characterize each tone differ, we show that the three cues obtained by fitting a parabola to the tone contour are an effective small set of cues such that any pair of tones is well distinguished by at least one of them. We propose using these three cues as a canonical choice for defining tone characteristics.

1pSCb45. The role of distributional information in cross-talker generalization of phonetic category retuning. Christina Tzeng and Lynne C. Nygaard (Emory Univ., 36 Eagle Row, Atlanta, GA 30322, ctzeng@emory.edu)

Listeners use lexical information to retune phoneme categories. Previous findings on the specificity of such perceptual adjustments suggest that cross-talker generalization occurs when talkers are acoustically similar. An alternative mechanism implies that listeners are sensitive to the statistical distribution of phonetic variation across speakers and can use this sensitivity to track both group- and talker-specific phoneme boundaries. Our objective is to disambiguate between these two possibilities. In the experiment, listeners hear words with an ambiguous fricative produced by multiple speakers of American English. Utterances across talkers vary in two ways: 1) the similarity of particular talkers' categorization curves and 2) the statistical distribution of sounds along the phonetic continuum (e.g., unimodal vs. bimodal) across talkers. We assess phonetic category retuning using novel words by novel talkers whose utterances either match or mismatch talker-specific properties or the distribution of phonetic realizations heard during exposure. If cross-talker generalization is similarity-based, listeners should generalize retuned phonetic representations to specific similar-sounding talkers at test. Alternatively, if listeners capitalize on the statistical distribution of exposure sounds, listeners should generalize to test talkers whose productions are consistent with the learned distribution. Results inform the conditions under which listeners form talker-independent representations of speech sound categories.

1pSCb46. Perceptual asymmetries in lexical tone perception. Ratree Wayland (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu) and Si Chen (Dept. of Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

Perceptual asymmetric relations between two stimuli such as perceptual confusions where stimulus A is more frequently confused as stimulus B than the reverse has previously been accounted for in terms of response bias due to the prevalent assumption that perceptual space is Euclidean (Polka & Bohn, 2003). Perceptual asymmetries in lexical perception have been reported for both infants and adults. Tsao (2008) found that a stimulus change from the background mandarin T1 (55) to the target Mandarin T3 (213) was easier than the reverse among one-year-old Mandarin learning infants. Politzer-Ahles *et al*. (2016) found that mismatch negativity (MMN) was attenuated among both native and non-native Mandarin listeners when Mandarin T3 was the standard and another deviant in comparison to the reverse. This study examines the effects of memory load and first language (L1 background) on perceptual asymmetry patterns in adult lexical tone perception. All possible pairings in both orders of Mandarin tones produced on two open syllables [ma] and [ni] will be presented to native Mandarin and American English listeners in an AX categorial discrimination task. To examine the effects of auditory memory load, two different inter-stimulus-interval (ISI), 250 ms and 1,000 ms, will be used.

1pSCb47. Non-native production of Mandarin T2/T3 pairs in disyllabic real and pseudo words. Xianghua Wu (California State Univ., Sacramento, 6000 J St., Sacramento, CA 95819, wu@csus.edu)

Mandarin rising (T2) and falling rising (T3) tones have been found to be the most confusing tone pair for native and non-native production due to similar F0 trajectory and equivalent lexical function known as T3 sandhi. This ongoing preliminary study investigates non-native production of T2 and T3 combinations in disyllabic real and pseudo words, aiming to examine whether T3 sandhi is realized solely at the real word level. Another goal of this study is to explore the effects of two distinctive orthographic systems in Mandarin (i.e., the sound-based pinyin and the meaning-based Chinese characters) on tone production. Ten native English speakers with 8-12 months experience in learning Mandarin participated in the production task. The stimuli were combinations of T2-T2, T2-T3, T3-T3, and T3-T2 grouped into four semantic and orthographic contexts: real and pseudo words in pinyin only, and those in both pinyin and Chinese characters. In the perception task, five native listeners of Mandarin identified the tone category for the first and second syllables. The results are expected to indicate the effects of tonal context, word meaning, and/or orthographic input on T2/T3 productions, and will be compared with earlier findings (e.g., Mok *et al*., 2018; Zhang *et al*., 2015).

1pSCb48. The acquisition of English stress by Kazakh-Russian bilinguals: The role of dominant language. Mahire Yakup (Lang., Linguist and Literatures, Nazarbayev Univ., Block 38-1104, 53 Kabanbay Batyr Ave., Astana, Aqmola 010000, Kazakhstan, yakefu.mayila@nu.edu.kz)

This study systematically investigates the cross-linguistic influence of the acquisition of English lexical stress by Kazakh-Russian bilinguals. Experiment 1 examined the Kazakh accent/stress pattern. Even though there is an argument about Kazakh stress, we used near minimal pairs in Kazakh such as balaDAR "children" vs. balaDAY "like a child, childish." By measuring duration, intensity, and pitch on 20 Kazakh-Russian bilinguals with strong Kazakh language level, we compared the stressed syllable versus unstressed syllables in the near minimal pairs and found duration as a stronger cue for the stress/accent in the syllables. In experiment 2 we investigated the roles of Russian as the dominant language in cross-linguistic influences. We used ten minimal pairs in Russian produced by 20 Russian bilinguals who claimed their native language is Russian. The results showed Kazakh-Russian bilinguals produced Russian stress using duration. Experiment 3 focuses on the acquisition of stress pattern in English by Kazakh-Russian bilinguals. We recruited 40 participants with (IELTS= 7.0) producing 10 English minimal pairs in three conditions. We found that duration and intensity are stronger cues than F0. None of the group used F0 as a stress cue in English lexical stress. The result will be discussed regarding dominant Russian language.

1pSCb49. Priming the representation of left-dominant sandhi words: A Shanghai dialect case study. 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), Yu-Fu Chien (Dept. of Chinese Lang. and Lit., Fudan Univ., Shanghai, Shanghai, China), and Jie Zhang (Dept. of Linguist, Univ. of Kansas, Lawrence, KS)

This study examines how the acoustic input (the surface form) and the abstract linguistic representation (the underlying representation) interact during spoken word recognition by investigating left-dominant tone sandhi, a tonal alternation in which the underlying tone of the first syllable spreads to the sandhi domain. We conducted an auditory-auditory priming lexical decision experiment on Shanghai left-dominant sandhi words, in which each disyllabic target ([tɕi55 dɛ31] "egg") was preceded by monosyllabic primes either sharing the same underlying tone ([tɕi55]), surface tone ([tɕi53] "machine"), or being unrelated to the tone of the first syllable of the sandhi targets ([tɕi24] "to remember"). Results showed a surface priming effect, but not an underlying priming effect. Moreover, the surface priming did not interact with speakers' familiarity ratings to the sandhi targets. The results are discussed in the context of how phonological opacity, productivity, and the directionality of tone sandhi patterns influence the representation of tone sandhi words as well as how the lexicality of the primes and the participants' usage pattern of Shanghai may have influenced the results.

1pSCb50. Effect of proficiency on perception of English nasal stops by native speakers of Brazilian Portuguese. Steven Alcorn, Irene Smith, Luis De La Cruz, and Rajka Smiljanic (The Univ. of Texas at Austin, 150 W. 21st St., Stop B3700, Austin, TX 78712, steven.alcorn@utexas.edu)

Brazilian Portuguese (BP) does not contrast /m/ and /n/ in coda position, leading to perceptual difficulties with minimal pairs like *lime*/*line* in L2 English. This study examined the effect of acoustic-phonetic and semantic enhancements on word recognition in noise for native English (n = 26) and L1 BP/L2 English (n = 21) listeners. They heard 120 high- and low-predictability sentences produced in clear and conversational speaking styles and mixed with speech-shaped noise. They typed the last word ending in either /m/ or /n/. The L2 listeners were separated into high- and low-proficiency (HP and LP) groups based on the perceived accentedness ratings of speech samples they provided. The results showed that native listeners benefited more from semantic than acoustic-phonetic enhancements alone, and the

accuracy was highest when both enhancements were combined. The HP L2 listeners recognized more target words overall than the LP group, and their benefit from semantic over acoustic-phonetic enhancements mirrored that of the native listeners. In contrast, LP listeners could not utilize the higher-level semantic information to the same degree in the absence of signal clarity. These findings show that L2 listeners benefited from acoustic-phonetic, semantic, and combined enhancements when recognizing /m/-/n/ coda pairs, but this effect was mediated by proficiency.

1pSCb51. Recall of clearly spoken sentences. Sandie Keerstock and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, 305 E. 23rd St. CLA 4.400 E9 Mail Code: B5100, Austin, TX 78712, keerstock@utexas.edu)

Recognition memory (i.e., identifying items as old or new) is higher for sentences spoken clearly than for sentences spoken casually. The clear speech benefit on recognition memory was observed for both native and non-native English listeners. The current study investigated the effect of clear speech on sentence recall, a more complex and effortful type of declarative memory that requires processing at phonological, lexical-semantic, morphosyntactic, and syntactic levels. Thirty native and 30 non-native English listeners heard 72 meaningful sentences (e.g., "The grandfather drank the dark coffee") produced in conversational and clear speaking styles by a female native English speaker. Participants were presented with 12 sentences blocked by speaking style, and instructed to memorize them. After each block of 12 sentences, participants wrote down what they remembered. Blocks were counterbalanced for speaking style. Responses were scored by number of keywords correctly recalled (verbatim memory) and sentence-level comprehension (gist memory). Preliminary results show enhanced verbatim memory for clear speech (i.e., higher rate of key words correctly recalled for sentences produced in clear speech) for both listener groups suggesting that hyper-articulated clear speech provides cognitive release and aids retention of surface form.

MONDAY AFTERNOON, 5 NOVEMBER 2018 SHAUGHNESSY (FE), 1:00 P.M. TO 5:25 P.M.

Session 1pSP

Signal Processing in Acoustics, Acoustical Oceanography, Architectural Acoustics, Musical Acoustics, Underwater Acoustics, and Noise: Machine Learning for Acoustic Applications II

Peter Gerstoft, Cochair
SIO Marine Phys Lab MC0238, Univ. of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238

Weichang Li, Cochair
Aramco Research Center - Houston, 16300 Park Row, Houston, TX 77084

Chair's Introduction—1:00

Invited Papers

1:05

1pSP1. Correlation of biological and acoustical measurements of the endangered Golden Cheeked Warbler with applications of machine learning. David P. Knobles (KSA LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept., The Univ. of Texas at Austin, Austin, TX), Lisa O'Donnell, Darrell Hutchinson, and William Reiner (Balcones Canyonlands Preserve, City of Austin, Austin, TX)

Acoustic measurements on *Setophaga chrysoparia* (Golden-cheeked warbler) have been made within the 40,000 acre Balcones Canyonland Preserve (BCP) in Austin, TX, where the long-term goal is to understand the potential effects of anthropogenic noise on breeding success. The anthropogenic sources of noise include road traffic, jet aircraft, helicopters, and urban development and utilization. During the breeding season from March through May, acoustical recordings were made in dense Ash Juniper forests for 12 hours per day at 12 locations within the BCP. The evolution of the number of songs of Type A and B and their variants are signatures of breeding success. The study attempts to apply principle component analysis for feature selection input into a feedforward neural network classifier of the A and B signatures, their variants, and the ability to automate the identification of individual birds. A feedforward neural network attempts to parameterize a mapping $y = f(x,\theta)$ for input features and category pairs of the A and B signatures and parameterization θ. Explored is the sensitivity of the number of hidden layers required to learn the optimal parameterization θ approximating f. [Work supported by City of Austin.]

1:25

1pSP2. Travel time tomography with local sparse modeling and dictionary learning. Michael J. Bianco and Peter Gerstoft (Marine Physical Lab., Univ. of California San Diego, Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92037, mbianco@ucsd.edu)

In this talk, we present a machine learning-based approach to 2D travel time tomography. Travel time tomography methods image slowness structures (e.g., Earth geology) from acoustic and seismic wave travel times across sensor arrays. Typically, slowness is obtained via an ill-posed linear inverse problem, which requires regularization to obtain physically plausible solutions. We propose to regularize this inversion by modeling rectangular groups of slowness pixels from the image, called patches, as sparse linear combinations of atoms from a dictionary. In this locally-sparse travel time tomography (LST) method, the dictionary, which represents elemental slowness features, is initially unknown and is learned from the travel time data using an unsupervised machine learning task called dictionary learning. This local model constrains small-scale slowness features, and is combined with a global model, which constrains larger-scale features with L2 regularization. In contrast to conventional regularization, which allows only for smoothness or discontinuous slowness, LST permits increased resolution where warranted by data. A *maximum a posteriori formulation* of LST is derived, which is solved as an iterative algorithm. LST performance is evaluated on both synthetic and real data in the context of ambient noise tomography.

1:45

1pSP3. The use of context in machine learning for bioacoustics. Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, marie.roch@sdsu.edu), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA), Danielle Cholewiak (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Erica Fleishman (Colorado State Univ., Fort Collins, CO), Kaitlin E. Frasier (San Diego State Univ., La Jolla, CA), Herve Glotin (Université de Toulon, La Garde, France), Tyler A. Helble (SPAWAR Systems Ctr. Pacific, San Diego, CA), John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA), Holger Klinck (Lab of Ornithology, Cornell Univ., Ithaca, NY), Scott Lindeneau, Xiaobai Liu (San Diego State Univ., San Diego, CA), Eva-Marie Nosal (Univ. of Hawai'i, Honolulu, HI), Kaitlin Palmer (San Diego State Univ., San Diego, CA), Yu Shiu (Lab of Ornithology, Cornell Univ., Ithaca, NY), and Gurisht Singh (San Diego State Univ., San Diego, CA)

Biological acoustic signals are often produced in context. Contextual information includes things such as timing between calls, conspecific or interspecies cues, and physical environmental cues such as sunrise or sunset. We show how some forms of contextual information can be used to improve the results of detection and classification tasks for biological acoustic signals. We examine how context can be used to improve labeled data, resulting in more-accurate classification results, as well as how learners can exploit context directly. We demonstrate these improvements via two bioacoustic detection/classification tasks. The first algorithm detects odontocete echolocation clicks. We used a decision support system that allowed analysts to label echolocation clicks using between call timing cues as well as other measurements and found that deep learners trained with these high quality data are able to detect clicks in adverse environments. The second algorithm applies contextual information surrounding North Atlantic right whale upcalls to improve precision and recall. [This work was supported by ONR Grant Nos. N00014-17-1-2867 and N00014-15-1-2299.]

Contributed Papers

2:05

1pSP4. Bayesian association of multiple infrasound events using long-range propagation models. Christophe Millet (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr), Michael Bertin (CMLA, ENS, Cachan, France), and Pierrick Mialle (CTBTO, Vienna, Austria)

The International Monitoring System (IMS) is a worldwide network of monitoring stations that helps to verify compliance with the Comprehensive Nuclear Test-Ban Treaty by detecting events that might indicate violations of the treaty. The IMS uses a combination of four technologies: seismological, radionuclide, hydroacoustic, and infrasound. An important limitation of these technologies is related to the fact that the structure of the propagation medium is partially unknown. This is especially true for the infrasound technology, and indeed, a current trend is to undertake the impact of atmospheric variability on waveforms using computational models. Simulation-based predictions, however, are inherently limited by large uncertainties. Further, many thousands of detections are recorded per week and thus, the problem of calculating plausible waveforms for subsets of detections often leads to computational demands that exceed available resources. In this paper, we present a new powerful statistical model for analyzing and interpreting large-scale IMS data. The method is based on a parallel Markov chain Monte Carlo algorithm and full-wave modeling. The method can detect association when multiple, interacting events are present in the data. The posterior probability of no association can be estimated, thereby providing a way to reduce the false alarm rate in operational-like environments.

2:20

1pSP5. A sensitivity-based Bayesian hierarchical process for calibrating infrasound propagation models. Christophe Millet (CEA, DAM, DIF, CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr)

Model calibration is viewed in the sense of adapting the full set of model parameters in order to get better resemblance between observations and major end-predictions. In this paper, we present a new probabilistic method to calibrate normal-mode-based propagation models using some observed data and sources of uncertainties. The unknown parameters are estimated using a multiple parallel Markov Chain Monte-Carlo method, for which convergence diagnostics are available. Using a few normal modes allows to rapidly estimate the statistical distributions of the arrival characteristics, on a mode-by-mode basis. In a sense, the unknown inputs "propagate" through the plausible waveguides with each mode and alters its amplitude and phase structure. The resulting waveform is obtained as a combination of individual wavepackets which depends continuously of the input parameters. Further, once the maximum likelihood has been identified, the reduced model can be extended to higher dimensions (with a larger number of modes) to better refine the calibration process. Numerical results are obtained using the FLOWS platform (*Fast Low Order Wave Simulation*), that integrate advanced spectral numerical methods and realistic representations of atmospheric disturbances. The method is used to revisit the infrasound signals recorded at I37NO during campaigns of ammunition destruction explosions at Hukkakero.**2:35**

1pSP6. Detecting infrasound signals and sources with dense network data. Michael A. Hedlin and Catherine de Groot-Hedlin (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0225, hedlin@ucsd.edu)

New, massive, datasets can be used to examine atmospheric phenomena in more detail than before but require analytical methods that are both efficient and capable of extracting useful information from faint signals immersed in noise. We have developed the AELUMA (Automated Event Location Using a Mesh of Arrays) method that recasts any dense network of sensors as a distributed mesh of triangular arrays. Each array provides a local estimate of signal properties. This information from arrays across the network is combined to estimate the source origin time and location. The process is repeated without oversight to catalog events. A key challenge in attributing signals to their source occurs when a large number of signals are detected nearly concurrently from different sources. We apply a cluster (decision tree) analysis that takes the results of array processing at all arrays to iteratively parse out subsets of detections from distinct sources. We used recordings of infrasound signals made at an extensive network of sensors to build a catalog of infrasonic activity across the continental United States. The accuracy of AELUMA is assessed using events for which the origin time and location are well known.

2:50

1pSP7. Graph clustering for localization within a sensor array. Peter Gerstoft (Noise Lab, Univ. of California San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu)

We develop a model-free technique to identify weak sources within dense sensor arrays using graph clustering. No knowledge about the propagation medium is needed except that signal strengths decay to insignificant levels within a scale that is shorter than the aperture. We then reinterpret the spatial coherence matrix of a wave field as a matrix whose support is a connectivity matrix of a graph with sensors as vertices. In a dense network, well-separated sources induce clusters in this graph. The support of the covariance matrix is estimated from limited-time data using a hypothesis test with a robust phase-only coherence test statistic combined with a physical distance criterion. The method is applied to a dense 5200 element geophone array that blanketed 7 km $\times$10 km of the city of Long Beach (CA). The analysis exposes a helicopter traversing the array.

3:05–3:25 Break

Invited Papers

3:25

1pSP8. Blind equalization and automatic modulation classification of underwater acoustic signals. Caitlyn N. Marcoux, Bindu Chandna, and Ballard J. Blair (The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, cmarcoux@mitre.org)

It is useful to passively characterize the underwater acoustic communications environment for a variety of purposes, including interference avoidance and enforcement of restrictions protecting marine mammals. Automatically determining the modulation of a received waveform can permit sonar or communications operations within the same bandwidth with a minimum of collisions, and it can identify a particular system operating outside its permitted regime. The characterization system needs to both determine the modulation and an unknown, time-varying channel impulse response since the transmitter and receiver are not coordinating. In this work, we demonstrate the use of blind equalization along with Convolutional Neural Networks for automatic classification of underwater signals. Our current research focuses on classification of constant modulus signals and demonstrates an approximate 30 percent improvement in modulation classification, compared to approaches without equalization, and a significant reduction in the amount of data needed for training. We considered BPSK, QPSK, MSK, FSK and 8-PSK modulations using simplified synthetic channels simulated via MATLAB to demonstrate our results. Future work is aimed at demonstrating classification improvement using realistic channel models simulated via the Sonar Simulation Toolkit, real underwater channels gathered from data collects, and additional underwater acoustic signal types.

3:45

1pSP9. Acoustic seafloor classification using machine learning and simulations of Helmholtz equations. Christina Frederick and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Blvd., Newark, NJ 07102, christin@njit.edu)

We discuss numerical methods for inverse problems in high frequency underwater acoustics, aiming to recover detailed characteristics of the seafloor from measured backscatter data generated from SONAR systems. The key to successful inversion is the use of accurate forward modeling capturing the dependence of backscatter on seafloor properties, such as sediment type, roughness, and thickness. Prediction models are often statistical models formed using years of collected ground-truth data or physical models that require costly simulations or unrealistic environmental assumptions. Sophisticated tools are needed to accommodate complicated scenarios, i.e., uncharted seafloor landscapes. To enable a rapid, remote, and accurate seafloor parameter recovery, we propose a combination of machine learning and high fidelity forward modeling and simulation of the physical wave propagation and scattering process. The idea is to partition environments on the order of kilometers in width into much smaller "template" domains, a few meters in width, where the sediment layer can be described using a few parameters. Machine learning is used to train a classifier using a reference library of simulations of Helmholtz equations on the domains. We investigate the potential of multilayer perceptrons and generalized regression in sediment identification using the reference library. [Work supported by NSF and ONR.]

4:05

1pSP10. Big data based ocean acoustic source localization via deep learning. Haiqiang Niu, ZaiXiao Gong, Haibin Wang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, nhq@mail.ioa.ac.cn), Emma Reeves Ozanich, and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Machine learning has shown its potential in ocean acoustic source localization when real data are available to construct the training data set (Niu *et al*., JASA 142, 1176–1188 (2017); Niu *et al*., JASA 142, EL455–460 (2017)). This paper investigates the cases suffering from deficient real observations, where big data generated from the acoustic propagation model are used as the training set and therefore

deep learning models could be exploited. To handle the environmental uncertainty, the multi-frequency acoustic data are generated using an acoustic propagation model (KRAKEN) based on tens of thousands of environments. A 50-layer residual neural network model (ResNet-50) is used to address this challenging problem. The performance on simulated and experimental data demonstrates that the proposed approach is able to adapt to a variety of environments in source localization problems.

Contributed Papers

4:25

1pSP11. Clustering analysis of crowd noise from collegiate basketball games. Brooks A. Butler, Mylan R. Cook, Kent L. Gee, Mark K. Transtrum, Sean Warnick, Eric Todd, and Harald Larsen (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

The relationship between crowd noise and crowd behavioral dynamics is a relatively unexplored field of research. Signal processing and machine learning (ML) may be useful in classifying and predicting crowd emotional state. As a precursor to performing ML, it is instructive to identify which crowd acoustic events an unsupervised ML algorithm would classify as unique. An initial set of audio features have been extracted from high-fidelity noise recordings of crowds at Brigham Young University men's and women's basketball games. A K-Means clustering analysis was conducted on half-second segments of the recordings using extracted features consisting of numerous statistical and spectral characteristics. For example, a clustering analysis performed on a one-twelfth-octave spectrogram of crowd noise recordings reveals there are approximately six unique events that occur during a game. The clusters for different audio feature sets are compared with human labeling of the different acoustical events. Implications for further ML algorithm development based on various sets of audio features are discussed.

4:40

1pSP12. Clustering analysis of inputs to a geospatial model of outdoor ambient sound. Brooks A. Butler, Kent L. Gee, Mark K. Transtrum, Katrina Pedersen (Phys. and Astronomy, Brigham Young Univ., 35 E 800 N, Provo, UT 84604, brooks.butler93@gmail.com), Michael M. James, and Alexandria R. Salton (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Outdoor ambient acoustical environments may be predicted through supervised machine learning using geospatial features as inputs. However, collecting sufficient training data is an expensive process, particularly when attempting to improve the accuracy of models based on supervised learning methods over large, geospatially diverse regions. Unsupervised machine learning methods, such as K-Means clustering analysis, enable a statistical comparison between the geospatial diversity represented in the current training dataset versus the predictor locations. In this case, the geospatial features that represent the regions of western North Carolina and Utah have been simultaneously clustered to examine the common clusters between the two locations. Initial results show that most geospatial clusters group themselves according to a relatively small number of prominent geospatial features, and that Utah requires appreciably more clusters to represent its geospace. Additionally, the training dataset has a relatively low geospatial diversity because most of the current training data sites reside in a small number of clusters. This analysis informs a choice of new site locations for data acquisition that maximize the statistical similarity of the training and input datasets. [Work funded by an Army SBIR.]

4:55

1pSP13. Optimal experimental design for machine learning using the Fisher information matrix. Tracianne B. Neilsen, Mark K. Transtrum, David F. Van Komen (Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), and David P. Knobles (KSA, LLC, Austin, TX)

Optimal experimental design (OED) refers to a class of methods for selecting new data collection conditions that minimize the statistical uncertainty in the inferred parameter values of a model. The Fisher information matrix (FIM) gives an estimate of the relative uncertainty in and correlation among the model parameters based on the local curvature of the cost function. FIM-based approaches to OED allow for rapid assessment of many different experimental conditions (e.g., input data type, parameterizations, etc.). In machine learning models, accurate parameter estimates are often not a priority (nor even desirable) as they have no direct physical meaning. Instead, one would like to minimize the uncertainty in the model predictions for several quantities of interest. FIM approaches to OED can be generalized to minimize statistical variance, not in parameters, but in predictions of the quantities of interest. This approach has been applied, for example, to systems biology models of biochemical reaction networks [Transtrum and Qiu, BMC Bioinformatics 13(1), 181 (2012)]. Preliminary application of the FIM to optimize experimental design for source localization in an uncertain ocean environment is a first step towards an efficient machine learning algorithm that produces results with the least uncertainty in the quantities of interest.

5:10

1pSP14. Curvature and distance amplitude correction for ultrasonic testing of a component with curved surface using deep learning method. Yujian Mei (Zhejiang Univ., 38 Zheda Rd., Hangzhou, Zhejiang Province 310027, China, meiyj1019@zju.edu.cn), Haoran Jin (Nanyang Technolog. Univ., Hangzhou, ZheJiang, China), Bei Yu, Eryong Wu, and Keji Yang (Zhejiang Univ., Hangzhou City, Zhejiang Province, China)

Surface curvature and distances from the transducer will have an effect on the amplitude of ultrasonic wave. When using the amplitude of ultrasonic signal to quantitatively size the flaws, a correction is necessary to assure all ultrasonic signal amplitude is based on the echo signal from flat surface. Hence, a deep learning method is proposed to adjust the signal amplitude which enters arbitrary curved surface. A deep learning model will adjust the signal amplitude as the radius of surface, distance from the transducer and attenuation coefficient of material. Then, the ultrasonic signal is imaged on the basis of corrected ultrasonic data. Compared to the ultrasonic imaging based on the raw data, the quantitative evaluation of flaw size is more precise by using the deep learning method. The deep learning model is trained by the CIVA simulation data. Ultimately, the model is verified by the comparisons between prediction and experimental data. The proposed method is adaptable to various curvature, distance, and material, which avoids manufacturing multiple standard specimens to satisfy lower costs and shorter cycle.

Session 1pUWa

Underwater Acoustics: Noise and the Environment

Gerald L. D'Spain, Chair
Marine Physical Lab, Scripps Institution of Oceanography, 291 Rosecrans St., San Diego, CA 92106

Contributed Papers

1:00

1pUWa1. Green's function emergence through cross-correlation of shipping noise. Emmanuel Skarsoulis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, N. Plastira 100, Heraklion GR-70013, Greece, eskars@iacm.forth.gr) and Bruce Cornuelle (Marine Physical Lab., Scripps Inst. of Oceanogr. UCSD, La Jolla, CA)

The cross-correlation of the underwater acoustic noise field at two locations is related to the time-domain Green's function (TDGF) between the two locations. The propagation conditions, the distribution of noise sources, and the hydrophone locations have a significant influence on the part of the TDGF that can be reliably reconstructed through the cross-correlation procedure. In the low-frequency part of the acoustic spectrum the dominating noise source is shipping, and for that case the effect of propagation conditions on the result of the cross-correlation procedure is studied by considering typical sound velocity profiles and hydrophone setups. The aim is to understand the effect that propagation conditions have on the cross-correlation procedure and point to the potential advantages of certain areas for the conduct of passive tomography experiments. Further, the effect of mooring motion on the cross-correlation result is addressed and ways to remove this effect, e.g., with the help of long-baseline navigation systems will be discussed. [Work supported by ONR.]

1:15

1pUWa2. Reductions in underwater radiated noise from shipping during the 2017 Haro Strait vessel slowdown trial. Alexander O. MacGillivray and Zizheng Li (JASCO Appl. Sci., 2305–4464 Markham St, Victoria, BC V8Z7X8, Canada, alex@jasco.com)

During 2017, the Vancouver Fraser Port Authority's Enhancing Cetacean Habitat and Observation (ECHO) program carried out a voluntary slowdown trial in Haro Strait (British Columbia) to investigate whether limiting vessel speeds to 11 knots would decrease noise in Southern Resident Killer Whale habitat. During the trial, JASCO collected source levels measurements on two underwater listening stations situated adjacent to the Haro Strait traffic lanes, while a third listening station in Georgia Strait measured noise from vessels outside the slowdown zone. Acoustic data from these three listening stations were analyzed using JASCO's PortListen® system, which tracks vessels using the Automated Identification System (AIS) and automatically measures the source levels of passing vessels, according to the ANSI standard for ship noise measurement (12.64-2009 R2014). The effects of voluntary slowdowns on vessel noise emissions were investigated, on a per-class basis, by comparing measurements of participating vessels with measurements obtained during control periods before and after the trial. Analysis of the trial data showed that speed reductions were an effective method for reducing broadband source levels for five categories of piloted commercial vessels: containerships, cruise vessels, vehicle carriers, tankers, and bulk carriers.

1:30

1pUWa3. An experiment of air-to-water sound propagation in deep water. GuangXu Wang (National Lab. of Acoust., Inst. of Acoust.,Chinese Academy of Sci., No.21st of Bei Si Huan Xi Rd., Beijing 100190, China, wgx@mail.ioa.ac.cn), LingShan Zhang, and ZhaoHui Peng (National Lab. of Acoust., Inst. of Acoust.,Chinese Academy of Sci., Beijing, China)

The underwater sound propagation from an airborne source has been paid much attention recently. To explore the underwater sound propagation from an airborne source in deep water, an experiment was conducted in South China Sea. A vertical line array of hydrophones with depth from 100 meters to 3400 meters was anchored on the seabed to detect the sound of a loudspeaker. Two microphones were placed under the loudspeaker to measure the source level. Both continuous and linear frequency modulation wave signals with frequency from 100 Hz to 2000 Hz were transmitted by the loudspeaker. The distances between the loudspeaker and the vertical line array were from 5km to 14.5km. Comparing signal detected by underwater hydrophones with signal detected by microphones in air, transmission loss of is figured out. The transmission loss predicted by a wave number integration model OAST is compared with the experimental results.

1:45

1pUWa4. Stochastic matched-field processing using polynomial chaos expansions to model environmental uncertainty. Roger M. Oba, Peter C. Mignerey, and Steven I. Finette (Acoust. Div., Code 7167, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, roger.oba@nrl.navy.mil)

A stochastic generalization of matched-field processing (MFP) is presented that incorporates stochastic steering *matrices* rather than steering *vectors*. This provides a natural framework for including environmental uncertainty associated with incompletely known environmental knowledge. Parametric probabilities allow use of polynomial chaos (PC) expansions to describe environmental uncertainty in a rigorous manner, yielding efficient representations of the stochastic steering matrices through the application of sparse sampling methods. In particular, PC methods apply to the uncertainty of sediment variability that can be modeled via horizontal spectral methods. With moderate variability over scales justified by data in the New Jersey shelf region, the sediment uncertainty can be modeled with relatively few PC expansion coefficients, which depend only on the mean amplitude of spectral components. The coefficients express the nonlinear dependence of the acoustic field on the sediment variability, essentially orthonormalizing higher moments. The coefficients lead immediately to stochastic steering matrices, which are then compared via various MFP processors against acoustic data for MFP source localization. [Work supported by the U.S. Office of Naval Research.]

2:00

1pUWa5. Underwater ambient noise spatial and temporal coherence at basin scale. G. Bazile Kinda, Florent Le Courtois, Yann Stéphan, Jean-Michel Boutonnier (Shom, HOM, 13, rue du Chatellier, Brest CS 92803, 29228, France, bazile.kinda@shom.fr), Jean-Yves Royer (Laboratoire Domaines Oceaniques, CNRS and Univ. of Brest, Plouzané, France), and Guilhem Barruol (Institut de Physique du Globe de Paris, Paris, France)

Underwater soundscape may be seen as an intrinsic feature of the environment in which it is recorded. In fact, underwater soundscape is a complex combination of sounds from a variety of acoustic sources, whether from natural or from human origin. When the distances between the sources and the receivers are sufficiently large, the received signals are strongly modified by the acoustic response of a continuously fluctuating marine environment. Then, spatio-temporal characteristics of the environment can be inferred from the measurements of soundscapes from hydrophone networks. The presentation aims to assess the spatial and temporal coherences of low frequency underwater ambient noise in the Indian Ocean basin. The data under consideration combine year round acoustic recordings from two recoding systems: on the one hand, data were from ~50 Ocean Bottom Seismometers (OBS) arranged in a network as large as 2000 km x 2000 km (http://www.rhum-rum.net/); and on the other hand acoustic data from hydrophones in the SOFAR channel.

2:15

1pUWa6. Sparsity-based shipping noise analysis and cancellation in underwater acoustic communication. Ruoyu Zhang (Commun. Res. Ctr., Harbin Inst. of Technol., 2 Yikuang St., Harbin 150080, China, 15B905028@hit.edu.cn), Lutz Lampe (Dept. of Elec. and Comput. Eng., The Univ. of Br. Columbia, Vancouver, BC, Canada), and Honglin Zhao (Commun. Res. Ctr., Harbin Inst. of Technol., Harbin, China)

We are interested in achieving reliable underwater acoustic communication (UWAC) in the presence of noise generated by vessel traffic, i.e., shipping noise. Our general approach is to determine an underlying structure in shipping noise and to exploit this structure to remove it from desired UWAC signals. Since a well-established statistical characterization of shipping noise is not available, in this paper, we discuss our evaluation of shipping noise data sets obtained from labeled recordings from an underwater acoustic hydrophone array deployed in the Strait of Georgia by Ocean Networks Canada. Motivated by the general observation that shipping noise exhibits a structure akin to impulse noise, we study to what extent shipping noise can be described as a sparse signal. Specifically, we apply the expectation-maximization method to train a Gaussian-mixture model with two components, in which the large variance component represents the impulse-noise part. In a second step, we use the model to perform noise cancellation in a multicarrier UWAC scenario. For this purpose, we apply a sparse-signal recovery formulation and a message passing algorithm for detection, i.e., a form of compressed sensing. Numerical results show the moderate level of mitigation of shipping noise interference achieved with the proposed method.

2:30

1pUWa7. Toward efficient shipping noise probability density function estimation using sea-lane source decomposition and probability theory. Florian J. Aulanier (Institut Maurice-Lamontagne, Fisheries and Oceans Canada, Pêches et Océans Canada 850, Rte. de la Mer, C. P. 1000, Mont-Joli, QC G5H 3Z4, Canada, florian.aulanier@dfo-mpo.gc.ca)

One goal of the Canadian Ocean Protection Plan (OPP) is to understand the potential effects of shipping noise on endangered whale species in order to mitigate them. Shipping noise environmental impact risk assessment requires the understanding of large scale, high-resolution time-space shipping noise distributions. The computation of such shipping noise probability density functions (pdf) requests considerable computing resources, especially when propagation occurs in complex and varying environments like shallow waters, canyons, or fjords. Besides, input parameters variability and uncertainties analyses require multiple hindcast, nowcast or forecast scenarios to be run when using a direct Monte-Carlo approach. In order to reduce the computation effort, sea-lane shipping traffic decomposition and probability theory are jointly used to derive shipping noise probability density functions with a logarithmic complexity algorithm, as opposed to the linear complexity of direct Monte-Carlo methods. First, a theoretical model is derived for straight shipping routes using simplified logarithmic propagation, validated with numerical examples, and used to perform a sensitivity analysis of shipping noise pdf to speed and route closest point of approach. The improvement in numerical efficiency is shown on a more realistic four sea-lane case scenario mimicking part of the summertime St. Lawrence estuary traffic. Eventually, *in situ* measurements will be available for comparison.

2:45

1pUWa8. Parallel processing and machine learning for long-timescale ambient noise measurements—Illustration with data from the Neptune Ocean Observatory offshore British Columbia. Amelie Klein (Mathematics, Univ. of Bath, University of Bath, Claverton Down, Bath, Avon and NE Somerset BA2 7AY, United Kingdom, ammk22@bath.ac.uk), Philippe Blondel (Phys., Univ. of Bath, Bath, Avon and NE Somerset, United Kingdom), and Kari Heine (Mathematics, Univ. of Bath, Bath, and NE Somerset, United Kingdom)

Seafloor observatories are a rapidly maturing technological approach, enabling the monitoring of marine soundscapes over large areas (basin scale) and long timescales (10+ years). However, the time needed to process broadband measurements, especially over large periods, often acts as a bottleneck. This is particularly true when combining multi-resolution analyses with assessing the impacts of relatively short transients. We are using parallel processing to enable machine learning approaches. To accelerate the computation of spectrograms, we have implemented a parallel processing method that uses the FFT algorithm FFTW3 (http://www.fftw.org/fftw3.pdf), using MPI/C++ on the High Performance Computing facilities at the University of Bath, and compared with spectrogram calculations from well-established software PAMGuide (Merchant *et al.*, Meth. Ecol. Evol. 2015), with Matlab's Parallel Computing Toolbox. This approach was tested on 1 month of broadband (96 kHz) measurements from the NEPTUNE node at Folger Deep. One month of data can be processed in <3 hours, to a dB accuracy even on short time segments, and that performance increases with the number of parallel processing units. Stability of the parallel approach has been tested with synthetic signals (e.g., chirps) and increasing signal-to-noise ratios. This enables much faster monitoring of long-term trends of important sound metrics.

3:00–3:15 Break

3:15

1pUWa9. Soundscape measurements and modeling of the Atlantic Outer Continental Shelf. James J. Murray (OASIS, Inc., 5 Militia Dr., Lexington, MA 02421, jjmurray@gmail.com), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA), and Richard L. Campbell (OASIS, Inc., Lexington, MA)

The impact of anthropogenic sound on the marine ecosystem has become an area of concern and subsequently increased research. As part of the Atlantic Deep Water Ecosystem Observatory Network (ADEON, adeon.unh.edu), a five-year collection effort of acoustic, oceanographic, marine ecosystem, and remote sensing satellite observations is underway. The centerpiece of the network is 6 long-term landers deployed off the coast of Jacksonville up past the mouth of the Chesapeake Bay. In this paper we present results from the first (of four) horizontal line array measurement cruises (deployed from a sailboat) between two of the moorings. The scientific objective is to evaluate the horizontal directionality of the ambient sound field and to measure the spatial correlation of the noise between the lander off of Wilmington NC and the lander off of the Virginia. A soundscape model, including environmental and source level variability will be presented. The acoustic sources will include ships, wind, rain, and marine mammals.

3:30

1pUWa10. Spatio-temporal modelling of underwater noise related to an arctic exploration drilling program. Jorge E. Quijano, David E. Hannay (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z 7X8, Canada, jorge.quijano@jasco.com), and Melanie Austin (JASCO Appl. Sci., Anchorage, AK)

One concern related to oil and gas industry activities in the arctic is the potential impact of anthropogenic noise on marine mammals. To gain understanding of the temporal evolution of the acoustic noise footprint from such activities, and to quantify the geographic extent at which sound levels remain within regulatory thresholds for marine mammal injury and behavioral disturbance, this work presents a comprehensive monitoring and modeling study of anthropogenic noise resulting from an exploration drilling project in the northeastern Chukchi Sea, conducted from July to October, 2015. Measured acoustic source levels of multiple vessels and drilling equipment were used to generate time-dependent wide-area noise fields. Model validation was carried out by comparing simulated results with high-resolution acoustic data acquired at multiple stations in the area. The study shows that noise radiating from up to 19 supporting vessels is usually dominant, but noise from certain drilling-related activities occasionally rises above vessel noise within a radius of 8 km from drillsites. Modeled scenarios also show that by constraining vessel positions to within a few kilometers of the drilling location, simultaneous construction of two oil well top holes results in minor increment of the aggregate acoustic footprint's extent relative to single-well drilling.

3:45

1pUWa11. Upslope propagation of low frequency deep ocean signals. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Kenneth Houston, Robert Tingley, Terry Nawara (Draper Lab., Cambridge, MA), Daniel Lawrence (Riptide Autonomous Solutions, Pembroke, MA), and Thomas Brovarone (Hydroacoustics, Inc., Henrietta, NY)

Over the period 18–21 March, 2017, 1-hour waveforms in the 60–120 Hz band were transmitted from a HLF-6A source deployed at 300 m depth in the deep northeast Pacific Ocean. These transmissions were recorded by two vertical hydrophone line arrays at various ranges in the deep ocean, GPS-equipped sonobuoys deployed on the continental shelf, and a bottom-mounted hydrophone in 900-m water at the western edge of Monterey Bay operated by the Monterey Bay Aquarium and Research Institute (MBARI). Although the source-receiver ranges to the sonobuoys and the MBARI hydrophone were approximately the same, the bathymetry profile up the continental slope to the sonobuoy location was significantly steeper than to the MBARI hydrophone. Only the upper part of the frequency band, above 90–95 Hz, is received with good signal-to-noise ratio, illustrating the high-pass temporal filtering of upslope propagation. Upslope propagation also acts as a low-pass spatial filter, allowing only lower-order modes to propagate onto the shelf. Numerical modeling is used to examine the predictability of the measured travel times and multipath arrival structure.

4:00

1pUWa12. Compressive sound speed profile estimation with direction-of-arrivals. Youngmin Choo (Defense System Eng., Sejong Univ., 209 Neungdong-ro, Gwangjin-gu, Seoul 05006, South Korea, ychoo@sejong.ac.kr) and Woojae Seong (Naval Architecture and Ocean Eng., Seoul National Univ., Seoul, South Korea)

Sound speed profile (SSP) predominates acoustic propagation in the ocean. In this work, the SSP is inverted using compressive sensing (CS) combined with direction of arrivals (DOAs) from beamforming. The travel times and the positions of the arrivals can be approximately linearized using their Taylor expansion with the shape function coefficients that parameterize the SSP. The linear relation between the travel times/positions and the shape function coefficients enables CS to reconstruct the SSP. The conventional objective function in CS is modified to simultaneously exploit the information from the travel times and positions. The proposed scheme is examined in the SWellEx-96 experimental environment.

4:15

1pUWa13. Ship speed reduction is an effective mitigation for underwater noise effects. Jason Wood (SMRU Consulting, PO Box 764, Friday Harbor, WA 98250, jw@smruconsulting.com), Dominic Tollit, and Ruth Joy (SMRU Consulting, Vancouver, Br. Columbia, Canada)

The Vancouver Fraser Port Authority's ECHO Program led a voluntary 2-month ship slowdown trial during which 56% of piloted ships slowed down to <13 knots in a 16 nautical mile corridor. The goal was to determine if a slowdown could be used as a mitigation measure to reduce ship related underwater noise effects in core Southern Resident killer whale habitat. A calibrated hydrophone system was used to measure ambient noise levels from 10 Hz to 100 kHz during the trial and a representative 2-month baseline period. The hydrophone was located 2.3 (inbound) and 5 km (outbound) from the center of the shipping lanes at 23 m depth. Analyses of data with ships present showed a median broadband noise reduction of 2.5 dB. This reduction was highest in the 10–100 Hz decade band (3.1 dB) and lowest in the 10,000–100,000 Hz band (0.3 dB). A statistical model found that received noise levels were best described by the distance to ships, the presence of small boats, water velocity, slowdown period, ship speed through water, and wind speed. This highlights that appropriate temporal scales and the inclusion of covariate data are needed to adequately measure ship related changes in underwater ambient noise levels.

4:30

1pUWa14. The preliminary study on spatial correlation of ocean sound field. Jin-bao Weng and Yan-ming Yang (Third Inst. of Oceanogr., State Oceanic Administration, No. 178 Daxue Rd., Siming District, Xiamen 361005, China, wengjinbao@tio.org.cn)

The sound field temporal correlation and spatial correlation, which are the foundation of the investigation of underwater signal space-time high-order characteristics, have important value in the underwater acoustic application. The spatial correlation is studied based on the shallow water acoustic propagation experiment data acquired in the northern South China Sea in 2017, and the deep water acoustic propagation experiment data acquired in the western Pacific in 2013. As for the explosive sound signals in shallow water, time domain waveform cross-correlation coefficients between signals from different propagation distance are calculated. In contrast, the linear frequency modulated signals in deep water need additional matched filtering. The signal processing results shows that, the overall spatial correlation is poor and the correlation radius is relatively small in shallow water, the convergence zone has an obviously better spatial correlation than the shadow zone for the deep water situation. The processing result is verified by simulation and analysis.

4:45

1pUWa15. The preliminary study on fluctuation of ocean sound field. Jin-bao Weng and Yan-ming Yang (Third Inst. of Oceanogr., State Oceanic Administration, No. 178 Daxue Rd., Siming District, Xiamen 361005, China, wengjinbao@tio.org.cn)

The ocean environmental variation causes the amplitude and phase fluctuations of acoustic signals that travel in the ocean acoustic waveguide. Sound propagation fluctuation is studied based on the deep water fixed-point sound propagation experiment data acquired in the northern South China Sea in 2016. In this experiment, two sets of submerged buoy systems, which integrate the emission and receiving of sound wave, have continuously and steadily worked as long as three months. Considering the Doppler frequency shift, the amplitude and phase of acoustic signals are calculated after matched filtering. The sound intensity scintillation index computed for the entire record is 0.8, which is less than the saturation value of one. In contrast, the scintillation index computed for the 2001 ASIAEX South China Sea Experiment is 2.6 and 1.7. The ocean environmental variation and its relation with sound propagation fluctuation are analyzed.

5:00

1pUWa16. Multi-shipping sources classification method based on the line spectral components of shipping radiated noise. Hongjian Jia and Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, jiahongjian_hrbeu@outlook.com)

The passive target detection based on shipping radiated noise is the key technology for underwater operations today. It has high research value both in civil and military applications. In this paper, the feature extraction of the underwater channels in the cepstrum domain is performed by using the line spectral components of the shipping radiated noise. The channel impulse response (CIR) function is used as the classification basis for the multi-shipping sources, and the line spectral components belonging to the same shipping source in the sonar received signal are searched to solve the classification problem of the multi-shipping radiated noise sources in the complex ocean background.

MONDAY AFTERNOON, 5 NOVEMBER 2018 OAK BAY 1/2 (VCC), 1:00 P.M. TO 4:45 P.M.

Session 1pUWb

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Variability in Shallow Water Propagation and Reverberation II

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

David R. Dall'Osto, Cochair
Acoustics, Applied Physics Laboratory at University of Washington, 1013 N 40th St., Seattle, WA 98105

Contributed Papers

1:00

1pUWb1. Excess attenuation caused by subducted bubbles at the ebb plume front in the Connecticut River. D. Benjamin Reeder, John E. Joseph, and Tarry Rago (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu)

A field experiment was carried out in the Connecticut River in June 2017, one goal of which was to investigate the low-to-mid-frequency acoustic propagation characteristics of the fresh water plume front outside the river mouth during ebb. Linear frequency-modulated (LFM) acoustic signals in the 500-2000 Hz band were collected during several tidal cycles. Transmission loss in excess of that which is expected by sound speed gradient-driven refraction is explored via acoustic propagation modeling. The excess attenuation at the ebb plume front is attributed to large clouds of subducted bubbles by downwelling at the ebb plume front.

1:15

1pUWb2. Temporal variability of propagation and reverberation during TREX13. Brian T. Hefner, Jie Yang, Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), and William S. Hodgkiss (UC San Diego, La Jolla, CA)

During the Target and Reverberation Experiment in 2013, a bottom-mounted, mid-frequency source and horizontal line array were deployed on the seafloor to measure reverberation over the course of the experiment. Extending from the source and parallel to shore was the "main reverberation track," a 7-km long and 1.5 km wide region where a majority of the acoustic and environmental measurements were made. Along this track, two vertical line arrays were deployed to continuously collect propagation data. The combination of the HLA and VLAs made it possible to make contemporaneous measurements of both transmission loss (TL) and reverberation level (RL) over a 3-week timespan. During this time, a storm passed through the site and the variability in the TL is found to be driven by different mechanisms. Prior to the storm, the variability was primarily driven by changes in the sound speed profile; after the storm, the water was well-mixed and the variability was a steady decay of TL while the rough sea surface decayed. This talk examines the variability in TL during these time periods and the associated changes in the measured RL. [Work supported by ONR.]

1:30

1pUWb3. Variability of target echo and clutter in the 2013 Target and Reverberation Experiment. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com), Jie Yang, Brian T. Hefner, and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Target and Reverberation Experiment was conducted in the Gulf of Mexico, off Panama City, Florida. Reverberation, noise, and target echo measurements were carried out for more than a month during April and May 2013, using a fixed source and fixed horizontal array receiver deployed in about 20 m of water. To support modeling efforts, a considerable number of ancillary environmental and transmission loss measurements were made. Various pulses in the 1.8–3.6 kHz frequency band were transmitted day and night, and have produced a rich data set. Initial results by various authors were published in the recent TREX13 Special Issue of the IEEE Journal of Oceanic Engineering. This work extends the analysis, concentrating on the variability of the echoes from various targets: a vertical hose, vertical arrays, a towed echo repeater, the hull of the towing ship, and other persistent bottom scatters, such as shipwrecks. In addition to the ping-to-ping variability of the echoes there are consistent trends, which are not understood, though probably due to oceanographic effects. An attempt will be made to relate the echo variability to measured and modeled reverberation and transmission loss. [Work supported by ONR, Ocean Acoustics, Code 22.]

1:45

1pUWb4. Characterizing acoustic propagation within the approach channel to Brandon Road Lock and Dam. Samuel L. Denes, David Zeddies (JASCO Appl. Sci., 8630 Fenton St., Ste 218, Silver Spring, MD 20910, sam.denes@jasco.com), and Christa Woodley (Engineer Res. and Development Ctr., US Army Corps of Engineers, Vicksburg, MS)

A propagation experiment was conducted in the approach channel to the Brandon Road Lock and Dam on the Des Plaines River in Joliet, Illinois, over the course of ten days in 2017. A sequence of frequency sweeps, continuous tones, and white noise was broadcast from an underwater transducer deployed from a shallow draft workboat at nine locations within the channel. A vertical array of one tri-axial particle motion sensor and three hydrophones was deployed from a movable platform and recorded signals throughout the approach channel. An additional bottom mounted, autonomous recorder with one hydrophone and one particle motion sensor was deployed outside of the navigable channel for the duration of the experiment. The engineered channel includes cement walls and floors near the steel lock doors, transitioning to natural surface shoreline down channel. For sources or receivers near the lock, the highly reverberant interfaces dominate propagation effects. This presentation will compare results of the measurement experiment with propagation models.

2:00

1pUWb5. Measured variability of acoustic propagation loss in the Salish Sea, British Columbia, Canada. Caitlin O'Neill (Fisheries and Oceans Canada, 9860 West Saanich Rd., Sidney, BC V8L 4B2, Canada, caitlin.v.oneill@gmail.com), Harald Yurk (Fisheries and Oceans Canada, West Vancouver, BC, Canada), and Svein Vagle (Fisheries and Oceans Canada, Sidney, BC, Canada)

The Salish Sea encompasses the waters between southern British Columbia mainland coast, Vancouver Island, and Washington State. Its waters are used by commercial and recreational fishers, host shipping traffic to the major ports of Vancouver and Seattle, and is travelled frequently by local ferries. It is also an important habitat for many marine mammal species, including the endangered Southern Resident Killer Whale (*Orcinus orca*). Southern Resident Killer Whales use calls, whistles, and clicks to communicate and find prey. Human activities on the water may affect the marine acoustic environment by masking whale vocalizations, which in turn will negatively impact the ability of the whales to forage and socialize. A field study was conducted to measure acoustic transmission loss at various places in the Salish Sea. Sound signals simulating killer whale call elements were broadcast from different angles to a calibrated hydrophone. Results from the study showed that acoustic transmission loss is highly spatially variable in this area. We suggest that these observations are due to changes in oceanography conditions and ambient sound levels, which are important factors to consider when assessing impact on Southern Resident Killer Whales. [Work supported by the Oceans Protection Plan of the Government of Canada.]

2:15–2:30 Break

2:30

1pUWb6. Confined space acoustic studies in Weymouth Fore River. Timothy F. Duda, Arthur Newhall, Peter Traykovski (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Michael B. Porter (Heat, Light and Sound Res., Inc., San Diego, CA), and Laurel Henderson (Heat, Light and Sound Res., Inc., La Jolla, CA)

An acoustic propagation study with two moving sources and three receiver positions was conducted in May 2018 in Weymouth Fore River, Massachusetts. Detailed bathymetric and geometric boundary information was previously collected. The data show strong inter-ping variability at ping rates of four Hz and one Hz. Signals were linear sweeps from 8-16 kHz and 21-34 kHz. The variation of the matched-filter envelope output, thought to be robust to phase alterations from small movements, was highly variable. This implies interfering multipath surpassing slowly varying response to boundary-interaction changes for the slowly varying source. The signatures of boundary interaction in models are compared to data to evaluate modeling procedures in the BELLHOP3D program. Interactions with concrete boundaries and steel ship and barge hulls are included in the analysis.

2:45

1pUWb7. A full-field time-domain model for reverberation in complex environments and waveguides with rough interfaces. Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A full-field perturbation approach [Ivakin 2016] has been used for modeling reverberation at given frequency in spatially varying layered environments and waveguides with rough interfaces. Average reverberation intensity was shown to be related with a local Bragg-wavenumber power spectrum of roughness through integration over the reference (unperturbed or smoothed) interfaces. The integrand includes a factor, a two-way propagation/interaction kernel, defined in terms of Green's function and a local contrast of acoustic parameters at the interfaces. This work extends the approach to time domain and includes additional integration over frequencies with another factor in the integrand, the radiated energy time-frequency distribution (ambiguity function). However, the integration within the signal frequency bandwidth can be significantly simplified using the frequency-range interference invariant. As a result, extremely fast estimations of reverberation time series (codas) can be made which require calculations of Green's function of the reference medium in the vicinity of the interfaces at only one (central) frequency. Numerical examples for time dependence of narrow- and broad-band reverberation in layered environments and waveguides with rough interfaces using PE-approximation for Green's function are presented. A possibility for remote monitoring of temporal and spatial variability of ocean interfaces based on propagation and reverberation measurements is discussed. [Work supported by ONR.]

3:00

1pUWb8. A study of underwater propagation model performance in complex environments with inherent uncertainty. Mark Langhirt (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mal83@psu.edu), Sheri Martinelli, and Charles W. Holland (Marine & Physical Acoust., ARL Penn State, State College, PA)

A number of computational models have been developed for purposes of modeling and simulation of acoustic wave propagation in underwater environments. Almost all of these models make approximations to the wave equation; furthermore, the method of implementation has a direct impact on the trade-off between accuracy and computational efficiency. Sonar technologies also vary greatly in range and frequency and one model is generally not suited for all applications; there are often subtle differences between them that can be highly dependent on the particulars of the environment. We investigate the performance of various models (ray tracing, level sets, normal modes, parabolic equation, and energy flux) using at-sea data from real world environments to assess their robustness of in the presence of the inherent uncertainty involved in reconstructing an underwater environment. In particular, we focus on the effect of perturbations in the sound speed profile. The sound speed profile, as a model input parameter, is subject to measurement error, as well as to undersampling of what, in many geographical areas of interest, is a complex spatial-temporal process. Ultimately, our objective is to establish some guidelines for practical use of the models in applications.

3:15

1pUWb9. Perturbation approximations for bathymetrically controlled ocean acoustic mode coupling. Brendan J. DeCourcy and Timothy F. Duda (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Iselin 3 (MS# 55), Falmouth, MA 02543, bdecourcy@whoi.edu)

A perturbation approximation of mode coupling equations for acoustic propagation in a submarine canyon environment is presented. The vertical modes and mode coefficients are represented by nearly adiabatic dominant mode functions and correction terms that are sensitive to the horizontal gradient of the variable bathymetry. This approach allows for a more accurate representation of the bottom boundary condition, and eliminates

computationally expensive implicit equations for the mode coefficients, replacing them with explicit equations for the nearly adiabatic contributions and the bathymetrically sensitive corrections. The presented method is applied to sloping bottom environments based on real bathymetric data to examine in detail the role of bottom variations in coupled mode propagation scenarios. [Work supported by the ONR.]

3:30

1pUWb10. Stage-wise orthogonal matching pursuit for estimation of time delay and Doppler doubly spreading of underwater acoustic channels. Feiyun Wu, Kunde Yang, Quan Sun, and Yunchao Zhu (Northwestern PolyTech. Univ., Youyi Rd. 127, Xi'an, Shaanxi 710072, China, wfy@nwpu.edu.cn)

The model of time delay and Doppler (DD) doubly spread is usually used for estimation of underwater acoustic channel (UAC) of time-varying and multi path. This study introduces the doubly spread model, then forms the optimization problem of channel estimations. By exploiting the sparsity of UAC, we propose the stage-wise orthogonal matching pursuit by inserting the orthogonal projection strategy on two domains. The traditional methods including least square (LS) based method, matching pursuit (MP), and orthogonal matching pursuit (OMP) methods are used for comparisons based on doubly spread models. Finally, the superiority of the proposed method is verified by experimental results.

3:45

1pUWb11. Geoacoustic inversion for bottom parameters in the deep-water area of the South China Sea. Shuanglin Wu, Zhenglin Li, and Jixing Qin (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring West Rd., Haidian District, Beijing 100190, China, wushuanglin@mail.ioa.ac.cn)

The properties of bottom acoustic parameters are greatly influenced on sound field prediction. Acoustic parameters in deep water are not well understood. We will discuss experience with geoacoustic inversion of transmission-loss (TL) data from 100 to 600 Hz from deep water regions. Bottom acoustic parameters are sensitive to the TL data in the shadow zone of deep water. Multiple steps TL inversion method is proposed to invertsound speed, density and attenuation in deep water. Both a uniform liquid half-space and a two-layer geoacoustic profile are assumed. Sound speeds of bottom are inverted by using TL data in the shadow zones at low frequency obtained in an acoustic propagation experiment conducted in the South China Sea (SCS) in summer 2014. Meanwhile, bottom densities are estimated combining with the Hamilton sediment empirical relationship. Attenuation coefficients at different frequencies are then estimated from the long range TL data by using the known sound speeds and densities as a constraint condition. The nonlinear relationship between attenuation coefficient and frequency is given in the end. The inverted bottom parameters can be used to forecast the transmission loss in deep water area of SCS very well.

4:00

1pUWb12. Extracting the intrinsic geo-acoustic parameters from the effective geo-acoustic parameters. Juan Zeng, Z. D. Zhao, Li Ma, and E. C. Shang (Inst. of Acoust., No. 21 Bei Si Huan Xi Lu, Beijing 100190, China, zengjuan@mail.ioa.ac.cn)

So far, most of the geoacoustic (GA) parameters are inverted through model-based approaches. The inverted GA parameters are not the intrinsic ones but the so-called "effective" ones. The "effective" GA parameters are model and frequency dependent and will be distorted due to model-mismatching. The real physical meaning of the term "effective" has not been fully addressed yet. In this paper we proposed to interpret the term of "effective" based on the Brekhovskikh's reflection principle, which reveals how the GA parameters influence the Green's function through the only channel, i.e. the bottom reflective coefficient. Based upon this point, we discuss the relationship between the "effective" GA parameters and the intrinsic GA parameters. First of all, it is shown that the real essential physical parameters, the reflective parameters, P(f) and Q(f) can be extracted from the "effective" GA parameters. Finally, with the help of P(f) and Q(f) some of the intrinsic GA parameters, the frequency dependence of the seabed attenuation and seabed sound speed profile can be retrieved.

4:15

1pUWb13. Range-dependent geoacoustic inversion for sound propagation in two seamounts environment. Lichen Pang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, lichenpang@163.com), Tao Hu, Licheng Lu, Qunyan Ren, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

This paper presents a work of range-dependent geoacoustic inversion for two seamounts environment with negative gradient sound speed profiles based on the transmission losses (TLs), collected in an experiment in the South China Sea, 2016. Bombs of 200 m and the mooring vertical line array (VLA) is applied for recording. Typical characteristics of shadowing and reflecting by seamounts are confirmed by the fluctuations of TLs with respect to range, indicating the complexity of sound propagation within seamounts environment. In the framework of matched field inversion (MFI), the geoacoustic model of two layers and the range-dependent acoustic model (RAM) is applied to produce the replica TLs, which is used to compute the cost function in combination with the collected TLs in terms of the least mean variance. The simulated annealing (SA) is expected to augment the searching efficiency of geoacoustic parameters. Results of four models with different parameters within 120km are obtained, with which the simulated TLs are in good accordance with experiment results for several frequencies and receiving depths. Conclusion is drawn that the effects of range-dependence of geoacoustic parameters in seamounts environment are as important as those of bathymetry on sound propagation, which should be seriously considered especially with negative gradient sound speed profiles.

4:30

1pUWb14. Comparison of velocity and attenuation measurements to the corrected effective density fluid model for gassy sediments. Fei Wang and Yiwang Huang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, wangfei0618@hrbeu.edu.cn)

To validate the corrected effective density fluid model for gassy sediments, an acoustic experiment was undertaken to measure the velocity and attenuation over the frequency range of 8 and 120 kHz in an indoor water tank. The velocity and attenuation obtained by different estimation methods have good consistency respectively. Both of them have four distinct peaks and there is one-to-one correspondence between velocity peaks and attenuation peaks. Through numerical simulation of the acoustic characteristics attributed to the resonance of gas bubbles with different radii, the existence of small bubbles causes the velocity to decrease in the vicinity of resonance frequency of large bubbles. Five modified Gaussian functions are introduced to model the bubble size distribution inside the gassy sediment. A best bubble size distribution is obtained by inversion of the velocity and attenuation data simultaneously, and the predictions of the corrected model are broadly in good agreement with the measurements.

Payment of an additional fee is required to attend this session.

MONDAY EVENING, 5 NOVEMBER 2018 SALON A (VCC), 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on An Introduction to Sound in the Sea

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

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. An introduction to sound in the sea. Thomas Dakin (Ocean Networks Canada, Univ. of Victoria, TEF-128A 2300 McKenzie Ave.,Victoria, BC V8W2Y2, Canada, tdakin@uvic.ca)

This introductory talk will address why underwater acoustics are important, its uses, how sound propagates so far in the ocean, man-made sounds, including shipping noise, and the impact of noise in the ocean. A short overview of organizations working on the issue of underwater noise will be given including those around the Salish Sea where the conference is being held. The way sound is measured and analyzed will be shown with local examples of marine life, earthquake and anthropogenic noise. A short exercise to show the effects of acoustic masking will be given, followed by an explanation of echolocation use by the Southern Resident Killer Whales and the implications of acoustic masking on finding food for this local and endangered species.

Session 2aAA

Architectural Acoustics: Architectural Acoustics and Audio: Even Better Than the Real Thing

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

Alexander U. Case, Cochair
Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Analysis of the effect on the indoor acoustic comfort with the living surfaces in the design studio. Filiz B. Kocyigit and Nazli N. Yildirim (Interior Architecture & Environ. Design, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

It is clearly shown in studies that acoustic comfort in the interiors play an effective role on directly increasing the productivity of users. Despite having the similar physical characteristics as the classrooms, design studios differ in terms of function and user behaviors. The training processes in the design studios have complicated function, but this process is limited in the classrooms. Recent research shows that living surface contribute to acoustical control at different frequencies, during the design periods at the studios. In these studies, living surfaces that provide indoor acoustic comfort are investigated in the general context. The effects of selected plants' structural differences on the indoor sound control at different frequencies are analyzed. Three different plants such as grass, moss, and aloevera are selected and their behaviors controlling the sounds of high, medium, and low frequencies in the indoor space are investigated. The behavior of these plants for the control of sounds in different frequencies in the studios has been examined. It is aimed to increase the educational efficiency of the users by using the combination of the most effective plant species in the studios' surface in terms of indoor acoustic comfort within the scope of the present data.

8:30

2aAA2. Sound system Déjà Vu all over again!—Revisiting past design approaches in an acoustically challenging multi-purpose space. Deb Britton and Dustin Goen (K2, LLC, 5777 Central Ave., Ste. 225, Boulder, CO 80301, deb@k2audio.com)

In the late 90s, we were tasked with designing a sound reinforcement system that would provide great speech intelligibility, in a highly reverberant space, without modifying any of the architectural finishes. The sound system had to support different types of events, held in different locations within the space, and with varying audience sizes. In 2015, we were asked to revisit the original loudspeaker design, and upgrade the sound system in that same space. Flash forward a few more years, and the entire building is getting a renovation—and we are once again designing a completely new sound system. This paper presents a case study of the upgrades, and describes the challenges, the lessons learned, and the changes in approach from the original upgrade through to the most recent design.

8:55

2aAA3. Beyond the chamber, plate, or digital reverb. Electronic architecture and the acoustically flexible recording environment. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Russ Berger (RBDG, Carrollton, TX)

For more than 20 years, Electronic architecture has been successfully integrated in a variety of venues for live performances. We will discuss how the inclusion of this technology in recording facilities has both altered recording methods, and expanded clientele at three recording facilities: The Dimenna Center for Classical Music is New York's premier facility for recording acoustic music. Recent projects include the soundtrack for the Coen Brothers' feature film *Hail, Caesar!;* the Grammy-nominated cast album for *The King and I* featuring Kelli O'Hara and Ken Watanabe; the soundtrack for James Schamus' *Indignation;* a Grammy Award-winning album by Room Full of Teeth; and an upcoming harmonia mundi release featuring Pablo Heras-Casado and Orchestra of St. Luke's. Sweetwater Music has recently expanded its recoding capabilities and upgraded the E-Coustic system with the latest hardware and software. Recent sessions include the Counting Crows, Adrian Belew, and Steve Curtis Chapman. Fo'Yo' Soul is Grammy award winning Kirk Franklin's new facility in Arlington Texas designed by Russ Berger. The studio (Uncle Jesse's Kitchen) features one of the first E-Studio product installations that allows acoustical conditions in the studio to be adjusted at the touch of a button, allowing soloists and ensembles to record in an environment tailored for comfort and creativity.

9:20

2aAA4. Audio rendering based on separation of direct and diffuse sound. Akio Ando (Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 3190 Gofuku, Toyama 930-8555, Japan, andio@eng.u-toyama.ac.jp)

Multichannel audio with many channels can capture the sound field with higher sense of reality. In the reproduction, however, a system with many loudspeakers can be setup only in the special room like a studio. Therefore, the audio rendering that converts the number of channels should be necessary in the reproduction. Many studies have been made on the audio rendering. These methods were based on the directional information of channels. On the other hand, there are two kinds of sense of reality: the reality by the object sound ("object reality") and that by the field sound ("field reality"). There are also two kinds of sound in the multichannel audio signal: the direct and diffuse sound. The direct sound brings the object reality, and the diffuse sound brings the field reality. Since the conventional rendering methods rely on the directional information, it may be difficult to handle the diffuse sound. In this paper, we address a rendering method based on the separation of direct and diffuse sound. Since the direct sound can be rendered by the conventional method, we clarify the requirements for the diffuse sound rendering from the viewpoint of sound reflection in the room.

9:45

2aAA5. Please turn it down!. Thomas J. Plsek (Brass/Liberal Arts, Berklee College of Music, MS 1140 Brass, 1140 Boylston St., Boston, MA 02215, tplsek@berklee.edu)

Sustained levels of 95 to 100 dBA and beyond are not uncommon in rock and pop music genres, leading to daily noise doses that are 100% or more above the recommendations established by the National Institute for Occupational Safety and Health. There are three options of addressing this situation: leaving the environment, using hearing protection, and turning the offending level down. It is the last of these that I will explore. The first two listed have problems with implementation. Using Leq measurements of various ensemble performance situations, the question addressed is whether or not musicians can learn to control the level of their instruments efficiently enough, referencing NIOSH recommended noise exposures, to make a meaningful difference in exposure time without significantly sacrificing the quality of the music.

10:10–10:25 Break

10:25

2aAA6. Effects for emphasis, enhancement, expression, and exaggeration. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Sound recordings built through multitrack production face a competitive soundscape. Signal processing strategies have evolved with the production goal of increasing the audibility of specific signal traits — making the subtle more apparent, the performance more compelling, the recorded art more successful.

10:50

2aAA7. AtatTATürk's Sarcophagus Acoustic Properties. Filiz B. Kocyigit (Architecture, Atilim Univ., incek, Ankara 06560, Turkey, filizbk@gmail.com)

This research investigates the acoustical characteristics of Atatürk sarcophagus. The Hall of Honor, which included the Atatürk's sarcophagus, was elevated from the whole mass, as it seemed strongly to the outside architect. The first floor is surrounded by this hall. The monument was built on a platform raised by six meters with a staircase in the direction of the square, with the ground closed and a massive wall with small windows, which was turned over with stone columns. The east entrance of the monument is at the beginning of the Aslan Road. At the beginning of this road, the entrance is strengthened by two guards, four meters high stairs are raised. When the monument is placed on the hill, two strong axes perpendicular to each other are taken as principal. Results, at the research, include equivalent sound pressure levels (Leq) as a function of location, frequency, and time of day. Special and important days like Revolution days are also researched. The spectra are generally flat over the 63–2000 Hz octave bands, with higher sound levels at lower frequencies, and a gradual roll off above 2000 Hz. Many units exhibit little if any reduction of sound levels in the nighttime.

11:15

2aAA8. Nature as muse: How cave acoustics can help us be more imaginative with our reverb choices. Yuri Lysoivanov (Recording Arts, Flashpoint Chicago, 28 N. Clark St. #500, Chicago, IL 60602, yuri.lysoivanov@columbiacollege.edu)

The cave is an oft-overlooked option for the audio practitioner's toolbox. With the aid of the science team from the National Park Service, we recorded impulse responses and analyzed the properties of several acoustically distinctive spaces inside Mammoth Cave in Kentucky, the longest cave network in the world. An understanding of the unique characteristics of these areas can help inform more creative approaches in reverb selection and educate us to be more innovative in music, sound, and room design.

Contributed Paper

11:40

2aAA9. Requirement of acoustical engineering discipline in building designs. Zohreh Razavi (Norman Disney & Young, 1166 Alberni St., Vancouver, BC V6E 3Z3, Canada, z.razavi@ndy.com)

While assurance of professional design and commitment for field review from all engineering disciplines and an architect is required, acoustical review is not mandated by legislations or By-laws. Although sound control review requirement is mandated by most By-Laws, this is at an architect's discretion to decide whether retaining an acoustical engineer for a sound control review for a project is required or not. Architects, even those focused on sustainable and human-centered design have often overlooked acoustical quality as a design goal. The launch of programs like the "WELL Building Standard" which includes several features related to acoustics, and addition of the "acoustic performance credit to LEED v4" are signs that the green building industry is starting to promote acoustic performance an integral component of sustainable design. Though architects are more aware of the importance of acoustics, achieving good acoustic performance is still overlooked. Once acoustical compliance letter is mandated during building permit, the importance of acoustics in building designs will be recognized.

TUESDAY MORNING, 6 NOVEMBER 2018 SHAUGHNESSY (FE), 9:00 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics: Topics in Animal Bioacoustics: Hearing

Dorian S. Houser, Chair
National Marine Mammal Foundation, 2240 Shelter Island Drive, San Diego, CA 92106

Contributed Papers

9:00

2aAB1. Impact of stimulus bandwidth on the estimate of the upper-frequency limit of hearing in toothed whales. Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Sean P. Coffinger (Univ. of California, San Diego, CA), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (U.S. Navy Marine Mammal Program, San Diego, CA), and Robert F. Burkard (Univ. at Buffalo, Buffalo, NY)

No consensus on the stimuli used in toothed whale hearing tests using evoked potential methods currently exists. However, stimulus bandwidth should directly affect determination of the upper-frequency limit (UFL) of hearing because of the steep reduction in hearing sensitivity at frequencies immediately below the UFL—the broader the stimulus bandwidth, the greater the amount of spectral "splatter" into lower frequencies where hearing is more sensitive. To test this hypothesis, variability in auditory evoked potential thresholds to tone-pips (with variable bandwidth) and amplitude-modulated tones was determined in dolphins near their UFL of hearing and ½ to 1 octave below where hearing is more sensitive. Subjects included both normal and hearing-impaired dolphins. At frequencies where hearing was sensitive and the audiogram was relatively flat, negligible changes in the hearing threshold with changing stimulus bandwidth were observed. Conversely, thresholds near the UFL of hearing declined with increasing stimulus bandwidth, regardless of whether dolphins had normal hearing or hearing impairment, and resulted in broadened hearing bandwidth estimates. Thus, it is recommended that stimulus bandwidth be reported in hearing tests where the UFL is tested since stimulus bandwidth affects thresholds through spectral splatter into lower frequency bands where hearing is more sensitive.

9:15

2aAB2. Forward masking recovery in bottlenose dolphins (*Tursiops truncatus*): Auditory brainstem responses to paired-click stimuli in high-pass masking noise. Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), Sean P. Coffinger (Psych. Dept., Univ. of California - San Diego, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Robert F. Burkard (Dept. of Rehabilitation Sci., Univ. at Buffalo, Buffalo, NY)

Adaptation is a fundamental process in the auditory system and underlies the automatic gain control system of echolocating bottlenose dolphins. As pulse-echo delay increases, auditory brainstem response (ABR) amplitudes to the echoes progressively increase. This study examined adaptation across cochlear frequency regions using a paired-click (i.e., forward masking) paradigm and high-pass masking noise. Bottlenose dolphins passively listened to paired click stimuli (20–160 kHz "pink" spectra). A "conditioning" click was followed by a "probe" click with equal amplitude and a time delay ranging from 125 to 750 μs. ABRs to click pairs were obtained with and without high-pass masking noise that precluded the basal turn of the cochlea from responding to the click stimuli. The ABR evoked by a single click (temporally aligned with the first click of the paired-click condition) was subtracted from the click-pair ABR to visualize the response evoked by the probe click. Probe ABR amplitudes recovered linearly with increasing delay relative to the conditioning click, and were approximately 70% of full response amplitude at 750-μs delay. Paired-click interval ABR amplitude recovery functions were similar for the unmasked and high-pass masked conditions. [Work sponsored by ONR.]

9:30

2aAB3. Responses of the auditory system of odontocetes to level changes of long lasting acoustical stimuli. Evgeniya Sysueva, Dmitry Nechaev, Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com), and Vladimir Popov (Inst. of Ecology and Evolution, Moscow, Russian Federation)

The effects of level changes of long lasting sound stimuli (tone pip trains) on evoked potentials (the rate following response, RFR) were investigated in a beluga whale (*Delphinapterus leucas*). The stimuli were of 64 kHz carrier frequency at levels from 80 to 140 dB re 1 μPa. During stimulation, the stimulus level either was kept constant (the steady-state stimulation) or was changed up/down by 20 or 40 dB every 1000 ms. After transition from a lower to upper stimulus level (increase), the response amplitude increased quickly and then decayed slowly. After transition from an upper to lower stimulus level (decrease), the response amplitude fell quickly and later recovered slowly. In the both cases, during the 1000 ms stimulus, the response amplitude almost reached that of the steady-state stimulus of the same level. The auditory system of the beluga (and, hypothetically, other odontocetes) may be characterized as quickly flexible and capable of quick adjustment of its responses to the current auditory scene. [This study was supported the Russian Science Foundation (Project No. 17-74-20107) awarded to E.V.S.]

9:45

2aAB4. Directional hearing sensitivity for 2–30 kHz sounds in the bottlenose dolphin (*Tursiops truncatus*). Alyssa W. Accomando, Jason Mulsow, Brian K. Branstetter (Biologic and BioAcoust. Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., #200, San Diego, CA 92106, alyssa.accomando@nmmpfoundation.org), James J. Finneran, and Keith Jenkins (US Navy Marine Mammal Program, Space and Naval Warfare Systems Ctr. Pacific, San Diego, CA)

Bottlenose dolphins (*Tursiops truncatus*) depend on sounds with frequencies <30 kHz for social communication and foraging, but little information on the directional dependence of hearing thresholds for these sounds exists. This study measured behavioral hearing thresholds for 2, 10, 20, and 30-kHz tones projected from eight different angular positions around the dolphin in both the horizontal and vertical planes to determine whether the receiving beam at these frequencies was directional. Omni-directional hearing was hypothesized for sounds below 30 kHz, but this hypothesis was rejected. Results from two bottlenose dolphins demonstrated a positive relationship between directivity and the frequency of the test tone, with asymmetric beam patterns. Directional hearing sensitivity declined most dramatically between 10 kHz and 2 kHz. The results suggest that dolphins' directional hearing is more pronounced for lower frequencies than previously predicted.

10:00

2aAB5. Passive echo detection and discrimination in bottlenose dolphins. Sean P. Coffinger (Psych., Univ. of California - San Diego, 10131 Caminito Zar, San Diego, CA 92126, sean.coffinger@gmail.com), Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Jason Mulsow (National Marine Mammal Foundation, San Diego, CA)

Echolocation in bottlenose dolphins (*Tursiops truncatus*) is an active process involving outgoing signal transmission and echo reception. Transmitted echolocation "clicks" interact with a target to produce echoes that contain representative information about the target. Echolocation allows dolphins to detect, discriminate, and identify complex targets using only their hearing. However, it is unclear whether dolphins capitalize on an internal representation of echolocation clicks for matched filter processing or whether the same information (and performance) can be obtained through passive echo reception (i.e., without a corresponding ensonifying click). Here, we investigate the passive echo discrimination ability of a bottlenose dolphin by embedding a target echo in a stream of distractor echoes with similar spectral and time-domain characteristics. Initial testing has demonstrated excellent echo-discrimination ability in the light of multiple distractor echoes of similar amplitude. Additional testing is underway with expanded distractor sets and echo-ordering complexity. The paradigm specifically investigates the plausibility of perceptual identification without the potential for matched filter processing. If representational identity is attainable via a passive process and not degraded relative to active echolocation, then the function of the echolocation click might not serve in matched filter processing, though it would remain important for ensonification and in determining target location.

10:15–10:30 Break

10:30

2aAB6. Dependence of bottlenose dolphin (*Tursiops truncatus*) auditory brainstem responses on noise burst rise time, amplitude, and envelope shape. Ryan A. Jones (Biologic and BioAcoust. Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, ryan.jones@nmmpfoundation.org), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), Jason Mulsow (Biologic and BioAcoust. Res., National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Dept. of Rehabilitation Sci., Univ. at Buffalo, Buffalo, NY)

A series of experiments examined the dependence of dolphin auditory brainstem response (ABR) peak amplitudes and latencies by stimulus level, rise times, and envelope shape. Stimuli were spectrally "pink" noise bursts with frequency content from 10 to 160 kHz, rise times that varied from 32 μs to 4 ms, and plateau sound pressure levels from 102 to 138 dB re 1 μPa. Cosine and cosine-cubed rise (and fall) envelopes were tested, and results were compared to those obtained for dolphins using noise bursts with linear rise envelopes [Finneran *et al*., J. Acoust. Soc. Am. **143**, 2914–2921 (2018)]. Results for the cosine and cosine-cubed envelopes showed that ABR peak amplitudes were a function of envelope sound pressure at the end of a fixed integration window of approximately 260 μs. This is congruent with the previous findings for linear envelopes. The ABR peak latencies for the cosine and cosine-cubed envelopes were dependent on the second derivative of the envelope pressure function, as opposed to the first derivative of the pressure function for linear envelopes. These data are consistent with single-unit and nearfield response data for terrestrial mammals. [Work supported by Navy Living Marine Resources and SSC Pacific Naval Innovative Science and Engineering Programs.]

10:45

2aAB7. The production and reception of underwater sound in Hawaiian monk seals (*Neomonachus schauinslandi*). Jillian Sills, Kirby Parnell, and Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, coll@ucsc.edu)

The endangered Hawaiian monk seal is a primitive phocid (true) seal endemic to the tropical Hawaiian Islands. At present, there is a lack of substantive bioacoustic information available for this species, with no formal descriptions of underwater vocalizations and limited data concerning underwater hearing. To address these knowledge gaps, we are working to better understand species-typical auditory capabilities and sound production by thoroughly evaluating a single individual living in human care. A mature male monk seal was trained to perform an auditory go/no-go signal detection task in water. Detection thresholds were measured for narrowband tones across the frequency range of hearing to generate a full underwater audiogram. Additionally, an acoustic recorder was placed in this monk seal's living enclosure for a full year, enabling characterization of his underwater repertoire and seasonal trends in vocal behavior. This study presents the first examination of underwater vocalizations in Hawaiian monk seals, provides insight into the perceptual abilities of this species and the evolution of underwater hearing within the phocid lineage, and enables improved assessments of noise effects on these vulnerable seals. [Work supported by Navy's Living Marine Resources Program.]

11:00

2aAB8. Bottlenose dolphin (*Tursiops truncatus*) vocal modifications in response to spectrally pink background noise. Maria Zapetis (Psych., Univ. of Southern MS, 118 College Dr., #5025, Hattiesburg, MS 39406, maria.zapetis@usm.edu), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), Carolyn E. Schlundt (Government IT Services / US Navy Marine Mammal Program, San Diego, CA), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), and Heidi Lyn (Psych., Univ. of Southern MS, Gulfport, MS)

The increase of oceanic shipping is a global predicament. The resulting proliferation of underwater noise levels is a serious concern for marine mammal welfare, as it has the potential to interfere with the communicative signals of bottlenose dolphins (*Tursiops truncatus*). The Lombard effect and other noise-induced vocal modifications may be employed to compensate for reduced signal-to-noise ratios. This study aimed to determine which vocal modifications dolphins use during experimentally controlled background noise conditions. Three dolphins (ages 30–52) participated in behavioral hearing tests using an adaptive up-down staircase, go/no-go procedure with 15 or 40 kHz tones. Tones decreased by 3 dB increments if a dolphin responded to the tone with a conditioned whistle, and increased by 3 dB if they did not. Dolphins performed this task during ambient noise (control) conditions, as well as three elevated bandpass noise (experimental) conditions: 0.6–5 kHz (115 dB re 1 μPa) and 0.6–10 kHz (115 and 125 dB re 1 μPa). The acoustic parameters of the dolphins' response whistles and victory squeals, such as duration, frequency, amplitude, and response latency, were analyzed and compared between control and noise conditions. These data provide a complement to field studies of odontocete noise-induced vocal modifications in the wild.

11:15

2aAB9. Stealthy moths avoid bats with acoustic camouflage. Thomas R. Neil, Zhiyuan Shen (Life Sci., Univ. of Bristol, Life Sci. Bldg., 24 Tyndall Ave., Bristol BS8 1TQ, United Kingdom, t.r.neil@bristol.ac.uk), Bruce W. Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), Daniel Robert, and Marc W. Holderied (Life Sci., Univ. of Bristol, Bristol, United Kingdom)

Intense predation pressure from echolocating bats has led to the evolution of a host of anti-bat defences in nocturnal moths. Some have evolved ears to detect the ultrasonic biosonar of bats, yet there are many moths that are completely deaf. To enhance their survival chances, deaf moths must instead rely on passive defences. Here, we show that furry morphological specializations give moth bodies and wing joints acoustic stealth by reducing their echoes from bat calls. Using acoustic tomography, echo strength was quantified in the spatial and frequency domains of two deaf moth species that are subject to bat predation and two butterfly species that are not. Thoracic fur determines acoustic camouflage of moths but not butterflies. Thoracic fur provides substantial acoustic stealth at all ecologically relevant ultrasonic frequencies, with fur removal increasing a moth's detection risk by as much as 38%. The thorax fur of moths acts as a lightweight porous sound absorber, facilitating acoustic camouflage and offering a significant survival advantage against bats.

11:30

2aAB10. The auditory attributes of Golden Eagles: Do Golden (*Aquila chrysaetos*) and Bald Eagles (*Haliaeetus leucocephalus*) share the same auditory space? Edward J. Walsh, Peggy B. Nelson, Julia Ponder, Christopher Milliren, Christopher Feist, Jeff Marr, Patrick Redig, and JoAnn McGee (Univ. of Minnesota, S39 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, ewalsh@umn.edu)

According to the U.S. Fish and Wildlife Service (2018), fatalities associated with wind turbine collisions have been reported for more than 200 bird species. Furthermore, based on statistical models of industry growth it has been suggested that as many as 1.4 million bird fatalities/year could be realized if the Department of Energy (DOE) wind energy goals are achieved; i.e., wind energy supplying 20% of total U.S. energy needs by 2030. Although passerine bird fatalities are most commonly reported, raptors that hunt by day, including bald and golden eagles, are the second most frequent casualties of turbine collisions. To address this concern, deterrence protocols designed to discourage eagles from encroaching into wind energy facility air spaces and thereby constrain the degree of risk to which birds are exposed are under investigation. As part of an effort to guide development of acoustic deterrence protocols, we report that the responsive frequency range of golden eagles is similar to that reported for bald eagles; upper and lower frequency limits of hearing are approximately 6.0 and 0.3 kHz, respectively. Suprathreshold response profiles measured in golden eagles exhibit standard features that will be compared with those of bald eagles. [Work supported by DOE grant #DE-EE0007881.]

11:45

2aAB11. Effects of numbers of repetitions, repetition rate, frequency separation, and frequency range on auditory streaming in budgerigars (*Melopsittacus undulatus*). Huaizhen Cai, Laurel A. Screven, and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, 206 Park Hall, Buffalo, NY 14228, huaizhen@buffalo.edu)

Auditory streaming has been widely investigated behaviorally and physiologically in animals. The paradigm used in European starlings (*Sturnus vulgaris*) by MacDougall-Shackleton and colleagues (1998, *J. Acoust. Soc. Amer.*) was used here in budgerigars (*Melopsittacus undulatus*) to measure the effects of different numbers of repetitions of HLH- (high and low tones) triplets, repetition rates, tone frequencies, and frequency separations between the two tones on auditory streaming. Similar to humans, budgerigars subjectively experienced the auditory streaming phenomenon; more repetitions of HLH- triplets, faster repetition rates, and larger frequency separations enhanced the streaming perception. Further, these results were consistent across the two frequency ranges used in this study. When increasing the numbers of HLH- repetitions, it took longer to establish the streaming perception in budgerigars, similar to the buildup phenomenon in humans. These results indicate, for the first time using a behavioral paradigm, that budgerigars experience auditory streaming in a manner similar to humans.

TUESDAY MORNING, 6 NOVEMBER 2018 ESQUIMALT (VCC), 8:00 A.M. TO 11:35 A.M.

Session 2aAO

Acoustical Oceanography, Animal Bioacoustics, Underwater Acoustics, and Signal Processing in Acoustics: Machine Learning and Data Science Approaches in Ocean Acoustics I

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Shima Abadi, Cochair
University of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011

Invited Papers

8:00

2aAO1. Data-driven discovery of dynamics for control. Steven Brunton (Univ. of Washington, 3026 NE 85th St., Seattle, WA 98115, sbrunton@uw.edu)

The ability to discover physical laws and governing equations from data is one of humankind's greatest intellectual achievements. A quantitative understanding of dynamic constraints and balances in nature has facilitated rapid development of knowledge and enabled advanced technology, including aircraft, combustion engines, satellites, and electrical power. There are many more critical data-driven problems, such as understanding cognition from neural recordings, inferring patterns in climate, determining stability of financial markets, predicting and suppressing the spread of disease, and controlling turbulence for greener transportation and energy. With abundant data and elusive laws, data-driven discovery of dynamics will continue to play an increasingly important role in these efforts. This work develops a general framework to discover the governing equations underlying a dynamical system simply from data measurements, leveraging advances in sparsity-promoting techniques and machine learning. The resulting models are parsimonious, balancing model complexity with descriptive ability while avoiding overfitting. The only assumption about the structure of the model is that there are only a few important terms that govern the dynamics, so that the equations are sparse in the space of possible functions. This perspective, combining dynamical systems with machine learning and sparse sensing, is explored with the overarching goal of real-time closed-loop feedback control of complex systems.

8:20

2aAO2. Machine learning applied to broadband sound propagation on the New England Shelf. David P. Knobles (KSA LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. Dept. and the Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mohsen Badiey (Elec. Eng., University of Delaware, Newark, DE)

Addressed is to what degree can variability of an ocean waveguide, such as that associated with the seabed, be inferred by estimating a mapping $y = f(x,\theta)$ for input/category pairs (x,y) and parameterization θ. Feedforward neural network algorithms are employed to learn the optimal parameterization θ approximating f. The mapping includes a chain of functions with hidden layers or functions not directly related to the input. Supervised learning is applied to broadband propagation in a shallow water environment called the New England *Mudpatch* whose acoustic characterization has been studied continuously since 2015. In 2017 about 400 MK-64 Signal Underwater Sound (SUS) and Combustive Sound Sources (CSS) were deployed in a 30×11 km^2 area with horizontal variability of the seabed for the purpose of inferring the statistics of the seabed characterization from the observed statistics of the acoustic field. The goal is to connect θ to physical source and waveguide parameter values. [Work supported by ONR.]

8:40

2aAO3. Estimation of the acoustic environment through machine learning techniques. Oscar A. Viquez, Erin M. Fischell, and Henrik Schmidt (Massachusetts Institute of Technol., 77 Massachusetts Ave., Bldg. 5-204, Cambridge, MA 02139, oviquezr@mit.edu)

The material composition of the bottom of shallow waterways can have significant effects on the corresponding acoustic environment, which autonomous underwater vehicles (AUVs) rely upon for sensing, navigation, and communication. Techniques that require an adequate environmental model are often used onboard AUVs to interpret the sensor data, but such techniques are often sensitive to even small deviations between the model and reality. A proposed approach to reduce this deviation is to use data from local soil and bathymetry surveys to generate environmental model approximations that may be loaded onto the vehicle in advance. During the early stage of deployment, the vehicle uses a K-nearest-neighbor classification approach to compare field calibration measurements with the various models, and select the most suitable solutions for use during the remainder of the active mission. Acoustic field simulations based on the environmental models are produced using normal mode theory as well as wavenumber integration, then compared with field array data. The techniques developed here could be used to facilitate the use of environment-sensitive approaches for detection and tracking during autonomous operations. [Work supported by the Office of Naval Research.]

9:00

2aAO4. Ocean acoustic range estimation in noisy environments using convolutional networks. Emma Reeves Ozanich, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu), Akshaya Purohit (Elec. and Comput. Eng., Univ. of California San Diego, La Jolla, CA), and Haiqiang Niu (Chinese Acad. of Sci., Inst. of Acoust., Beijing, China)

Machine learning has been able to accurately estimate ocean acoustic source range in real data using passive ship signals (Niu *et al.,* JASA 142, 1176–1188 (2017); Niu *et al.,* JASA 142, EL455–460 (2017)). In this paper, we train a convolution neural network (CNN) to learn range-dependent features directly from the multi-frequency complex pressure field. The CNN network uses 2D inputs to take advantage of spatial relationships between frequency and depth. KRAKEN is used to simulate pressure magnitude and phase received on a vertical array at multiple frequencies, ranges, and signal-to-noise ratios (SNR), generated by adding white Gaussian noise. We show that performance at low SNR can be improved by increasing the size of the training data. The CNN-LSTM (long short-term memory) network, which incorporates sequential learning, is compared to the CNN at varying SNR. The models are tested on real data from the SBCEx17 experiment.

Contributed Papers

9:20

2aAO5. Classification of multiple source depths in a time-varying ocean environment using a convolutional neural network (CNN). Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

Machine learning (ML) is the idea that there are generic algorithms that can tell us something interesting about a set of data without us having to write any custom code specific to the problem. Instead of writing code, we feed data to the generic algorithm and it builds its own logic based on the data. In this paper, ML is applied to classification of multiple (4) sources at various depths in a shallow-water waveguide where the input data are time-varying channel impulse responses (CIRs) received by a vertical array (4 hydrophones) from each of the sources over a period of 22 hours (528 examples). Specifically, the first 16 hours of CIRs are used for training a 4-layered CNN (convolutional neural net) to exploit the structure of CIRs in the two-dimensional time and depth domain, similar to image processing. Our hypothesis is that the dataset collected over a semi-diurnal tidal period (>12 hours) could capture the internal representations or feature vector needed for classification. The trained model is then tested on the next 6 hours of CIRs and achieves 98% accuracy, indicating the potential of ML in underwater applications.

9:35

2aAO6. Estimating the probability density function of transmission loss in an uncertain ocean using machine learning. Brandon M. Lee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu)

Predicted values of transmission loss (TL) in ocean environments are sensitive to environmental uncertainties. The resulting predicted-TL uncertainty can be quantified via the probability density function (PDF) of TL. Monte Carlo methods can determine the PDF of TL but typically require thousands of field calculations, making them inappropriate for real-time applications. Thus, a variety of alternative techniques based on polynomial chaos, field shifting, modal propagation in ocean waveguides, and spatial variations of TL near the point(s) of interest have been proposed. This presentation describes an innovative approach to estimating the PDF of TL based on nominal TL, ocean environmental parameters, and machine learning. This approach has two main challenges. First, appropriate representations must be found for *ground-truth* PDFs of TL generated from Monte Carlo calculations so that a neural network can be constructed to predict each parameter of the estimated PDF of TL. Four such representations are considered here. And second, a framework must be developed to generate training data for the neural networks. A proposed framework that predicts candidate environments' training utility without computing any PDFs of TL is described. The performance of this approach is analyzed and compared to that of prior techniques. [Sponsored by ONR.]

9:50–10:05 Break

10:05

2aAO7. Using machine learning in ocean noise analysis during marine seismic reflection surveys. Shima Abadi (Eng. and Mathematics, Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, abadi@uw.edu)

Marine seismic reflection surveys use repetitive broadband sound signals to image the structure of the seafloor. High energy sound signals generated by airguns are recorded by single or multiple long horizontal hydrophone arrays towed behind the vessel. Marine seismic reflection surveys raise concern over their effects on marine animals. In particular, there is concern about (1) the impact of the airgun sound power level on marine mammals and (2) the increase in the background noise level caused by the reverberation between shots which may mask marine mammals' vocalizations and degrade the performance of the passive acoustic monitoring systems. Quantifying these effects is a challenging task due to the complexity of local geology, seafloor topography, and uncertainty in water properties. In this study, a machine learning approach is used to predict the airgun sound power level and the reverberation level off the ocean floor. The data utilized in this study are from the COAST (Cascadia Open-Access Seismic Transects) seismic reflection survey conducted with the R/V Marcus Langseth in July 2012. This experiment spanned a wide range of water depths from the continental shelf to deep water. Data was recorded by an 8-km horizontal array with 636 hydrophones sampled at 500 Hz.

10:20

2aAO8. Underwater acoustic target recognition using graph convolutional neural networks. Razi Sabara and Sergio Jesus (Univ. of Algarve, Gambelas Campus, Centro de Investigacao Tecnologica do Algarve, Faro 8005-139, Portugal, razi.sabara@gmail.com)

Motivated by recent progress in signal processing on graphs and convolutional neural networks, we have developed an underwater acoustic target recognition system based on graph convolutional neural networks. We evaluate our framework by application to various real-world datasets and validate its effectiveness. Our experiments demonstrate that the proposed approach achieves high classification accuracy.

10:35

2aAO9. Exploring matrix and tensor factorization for discovering latent structures in large echosounder datasets. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu) and Valentina Staneva (eSci. Inst., Univ. of Washington, Seattle, WA)

Moored autonomous echosounders are increasingly popular as an integrated component in ocean observing systems for measuring biological response to environmental changes. Different from ship-based observations, these large echo datasets often lack concurrent ground truth information from net or optical samples and reliable calibration, both of which are required for conventional analysis routines. However, even though accurate inference of organism abundance is not possible in such scenarios, rich spatio-temporal structures exist in the data and may be exploited to capture variation of migration patterns of different animal groups in the water column. Here, we explore the use of unsupervised matrix and tensor factorization approaches to analyze the long-term echo data sets from the Ocean Observatories Initiative (OOI) network. The data stretch in multiple dimensions, including time, space, and frequency. We restructure the echo data to explore latent structures at different temporal scales using different factorization formulations. Outputs from different methods are compared among one another and with those from conventional echo analysis routines. The results show the importance of augmenting generic factorization formulations with temporal and spatial continuity constraints for biologically meaningful analysis of large echosounder datasets.

10:50

2aAO10. Navigating noise when comparing satellite and acoustic remote sensing data. Carrie C. Wall, Kristopher Karnauskas, Maxwell B. Joseph, Joseph McGlinchy, and Brian R. Johnson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu)

Using a novel combination of NASA satellite products and acoustic remote sensing measurements, we aim to examine the link between the surface expression and vertical structure of ocean productivity and biomass in the California Current System. Water column sonar data collected by the National Marine Fisheries Service for fisheries management have been archived at the NOAA National Centers for Environmental Information. These data are used to evaluate the variability of the marine biogeography. To identify patterns of acoustic reflectance of marine organisms, the data must first be free of noise and seafloor acoustic returns. However, removing the variety of noise present has proved to be a challenge due to need for

manual tuning, a resource prohibitive step, and the size and complexity of the acoustic data. Collaboration with the University of Colorado's Earth Lab has led to the development of machine learning models to automatically remove noise from this large dataset. An analysis of the fully automated, semi-manual, and machine learning quality control processes will be presented. The cleaned acoustic data are then compared to the interannual variability of the distribution of surface chlorophyll concentration and temperature from satellite ocean color measurements.

11:05

2aAO11. Data assimilation for oceanographic and acoustic forecasting. EeShan C. Bhatt and Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., Rm. 5-223, 77 Massachusetts Ave., Cambridge, MA 02139, eesh@mit.edu)

Data assimilation relates observed physical data with dynamic modeling to provide a joint estimate of a field of interest that is bounded by input error. Here, a modular data assimilation framework is introduced for acoustic and sound speed field estimation with a discussion on streamlining separate and varied oceanographic data sources into previously established four-dimensional acoustic modeling. This methodology is applied to acoustic and physical data from an experiment in the Santa Barbara Channel, where passing cargo ships served as sources of opportunity, and a sensitivity analysis for acoustic transmission due to changing sound speed fields is presented. [Work supported by ONR under the Information in Ambient Noise MURI.]

11:20

2aAO12. Source localization using a compact tetrahedral array. 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), Aditi Tripathy (Ocean Eng., Univ. of Rhode Island, Kingston, RI), Makio Tazawa (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer Amaral, Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Ying-Tsong Lin, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

We localized sound sources collected on a compact tetrahedral hydrophone array in a continental shelf environment south of Block Island, Rhode Island. The tetrahedral array of phones, 0.5 m on a side, was deployed to monitor the underwater sound of construction and operation of the first offshore wind farm in the United States. Signals from shipping and marine mammals, including fin whales, humpback whales, and right whales, were detected on the array. Directions of arrival (DOAs) for a number of signals were computed using a time difference of arrival technique. Given the DOAs, ranges were estimated using supervised machine learning techniques outlined by Niu *et al.* (JASA, 2017). The approach was tested using simulated data from Kraken assuming environmental information consistent with this continental shelf environment. Performance on signals from individual ships and marine mammals is presented. Ship localizations are compared to Automated Identification System (AIS) fixes. An error analysis is also presented. [Work supported by the Office of Naval Research and the Bureau of Ocean Energy Management.]

TUESDAY MORNING, 6 NOVEMBER 2018 COLWOOD 1/2 (VCC), 8:00 A.M. TO 11:50 A.M.

Session 2aBAa

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

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

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Chair's Introduction—8:00

Invited Papers

8:05

2aBAa1. Picosecond opto-acoustics for the remote ultrasonography of single cells. Bertrand Audoin (Université Bordeaux, 341 cours de la Libération, Talence 33405, France, bertrand.audoin@u-bordeaux.fr)

Picosecond ultrasonics (PU) is a technique that offers the possibility to generate and detect acoustic waves remotely with an unequalled frequency bandwidth. After decades of developments and applications for researches in solid-state physics, a vast application field was recently demonstrated for PU in biology and medicine. The technique allows imaging single cells with a lateral resolution limited by optics and with the mechanical properties as the contrast mechanism. The opto-acoustic images can reveal the structure of the cell nucleus, the fine details of the actin network and of the adhesion pattern. In addition to imaging capabilities, quantitative analysis of the interaction of the GHz acoustic waves with the complex cell medium provides information on the cell nanostructure. For instances, the standard deviation of impedance data we measured for single nuclei revealed differences between different cell types arising from the multiplicity of local chromatin conformations within the nucleus. Moreover, the distribution of the cell-substrate interface stiffness

we identified allowed us to separate the contribution of passive and active adhesion processes. The technique allows thus gaining new insights into cell mechano transduction.

8:25

2aBAa2. Cell quake elastography. Guy Cloutier, Pol Grasland-Mongrain (Univ. of Montreal Hospital Res. Ctr., 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada, guy.cloutier@umontreal.ca), Ali Zorgani (Univ. of Lyon, Bron, France), Shoma Nakagawa, Simon Bernard, Lia Gomes Paim, Greg FitzHarris (Univ. of Montreal Hospital Res. Ctr., Montreal, QC, Canada), and Stefan Catheline (Univ. of Lyon, Lyon, France)

The ability to measure the elasticity of a cell provides information about its anatomy, function, and pathological state. Many techniques have been proposed to measure mechanical properties of single cells but most need a model of the cell characteristics and the fixation of the cell on a substrate. Moreover, current measurements take seconds to hours to perform, during which biological processes can modify the cell elasticity. Here, we have developed an alternative technique by applying shear wave elastography to the micrometer scale. Elastic waves were mechanically induced in live mammalian oocytes using a vibrating actuator. These audible frequency waves were observed optically at 205,000 frames per second and tracked with an optical flow algorithm previously developed in the context of ultrasound elastography. Whole-cell elasticity was then reconstructed using an elastography method inspired by the seismology field. Using this approach we showed that the elasticity of mouse oocytes is decreased when the oocyte cytoskeleton is disrupted with cytochalasin B. The technique is fast (less than 1 ms for data acquisition), precise (spatial resolution of a few micrometers), and able to map internal cell structures. The proposed method may represent a tractable option for interrogating biomechanical properties of diverse cell types.

8:45

2aBAa3. Acoustic propagation in the complex sclera for understanding the biomechanical changes associated with myopia. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Sally A. McFadden (Vision Sci., Hunter Medical Res. Inst. and School of Psych., Faculty of Sci., Univ. of Newcastle, Newcastle, NSW, Australia), and Quan V. Hoang (Singapore Eye Res. Inst., Singapore National Eye Ctr., DUKE-NUS, Singapore, Singapore)

Myopia affects up to 2.3 billion people and, although minimal levels of myopia are considered a minor inconvenience, high myopia is associated with sight-threatening pathology in 70% of patients. The anatomic changes occurring in the posterior sclera play an important role in myopia progression. Therefore, we investigated the biomechanical properties of the sclera using a well-established myopia model in guinea pigs. We employed two ultrasound-based approaches to better understand and quantify the microstructural changes occurring in the posterior sclera associated with high-myopia development. The first approach applied quantitative-ultrasound (QUS) methods to intact *ex-vivo* eyeballs of myopic and control eyes using an 80-MHz ultrasound transducer. The second approach used a scanning-acoustic-microscopy (SAM) system operating at 250 MHz to form two dimensional maps of acoustic properties of thin sections of the sclera with 7-μm resolution. We tested the hypothesis that the QUS and SAM-based properties are altered in myopia and can provide new contrast mechanisms to quantify the progression and severity of the disease as well as determine where the posterior sclera is most affected. Ultimately these methods will provide novel knowledge about the microstructure of the myopic sclera that can improve monitoring and managing high-myopia patients.

Contributed Paper

9:05

2aBAa4. Analytical solution for elliptic shear standing wave pattern in a bounded transversely isotropic viscoelastic material. Martina Guidetti and Thomas J. Royston (BioEng., Univ. of Illinois at Chicago, 851 South Morgan St., MC 063, Chicago, IL 60607, troyston@uic.edu)

Dynamic elastography methods—based on photonic, ultrasonic, or magnetic resonance imaging—are being developed for quantitatively mapping the shear viscoelastic properties of biological tissues, which are often altered by disease and injury. These diagnostic imaging methods involve analysis of shear wave motion in order to estimate or reconstruct the tissue's shear viscoelastic properties. Most reconstruction methods to date have assumed isotropic tissue properties. But, application to tissues like skeletal muscle and brain white matter with aligned fibrous structure resulting in local transverse isotropic mechanical properties would benefit from analysis that takes into consideration anisotropy. A theoretical approach is developed for the elliptic shear wave pattern observed in transverse isotropic materials subjected to axisymmetric excitation normal to the fiber axis. This approach, utilizing Mathieu functions, is enabled via a transformation to an elliptic coordinate system with isotropic properties and a ratio of minor and major axes matching the ratio of shear wavelengths perpendicular and parallel to the plane of isotropy in the transverse isotropic material. The approach is validated via finite element analysis cases studies. This strategy of coordinate transformation to equivalent isotropic systems could aid in analysis of other anisotropic tissue structures. (Grant support: NIH AR071162.)

Invited Paper

9:20

2aBAa5. Mechanical biomarkers by torsional shear ultrasound for medical diagnosis. Guillermo Rus, Juan M. Melchor, Inas Faris, Antonio Callejas, Miguel Riveiro, Francisca Molina, and Jorge Torres (Structural Mech., Univ. of Granada, Politecnico de Fuentenueva, Granada 18071, Spain, grus@ugr.es)

The WHO estimates that 15 million babies yearly (1 in 10) will be born preterm. Worldwide, complications of preterm births have supplanted pneumonia as the primary cause of child mortality [1]. The biology of cervical ripening that leads to birth is poorly understood, and there is no clinical tool to quantitatively evaluate the cervical biomechanical state, which in words of Feltovich [2]"… likely contributes to the reason the singleton spontaneous preterm birth rate has not changed appreciably in more than 100 years." Towards this problem, we work on enabling new sensor technologies sentient to soft tissue biomechanics, to endow a new class of biomarkers that quantify the mechanical functionality of the cervix, and indeed any soft tissue, ranging pathologies from tumors, atherosclerosis, liver fibrosis to osteoarticular syndromes. Ultrasonic characterization of soft tissue has been developed as a clinical diagnostic tool [3] and evolved through different technologies including our torsional wave principle [4]. Our recent advances covering (a) torsional waves (shear elastic waves that propagate in quasifluids radially and in depth in a curled geometry), (b) sensors (based on a novel arrangement of concentric sandwiches of elements), (c) propagation models and (d) patient testing, are allowing to quantify the mechanical functionality beyond linear parameters: dispersive and nonlinear. [1] WHO, 2012; [2] Feltovich. *AJOG*207(2012):345–354 [3] Barr. *Ultras Quart*28(2012):13–20; [4] Melchor. *Ultrasonics*, 54(2014):1950–1962.

Contributed Papers

9:40

2aBAa6. Shear wave elastography with combined high spatial and temporal frequency excitations. Matthew W. Urban and Piotr Kijanka (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Shear wave elastography is a method for quantitatively measuring the mechanical properties of soft tissues. Shear wave attenuation is high in soft tissues and increases with frequency. In a recent work, high resolution and accuracy can be reached at higher temporal frequencies. To combat the high shear wave attenuation and to obtain higher accuracy and lower variability, we propose to combine acoustic radiation force excitations that have multiple push beams and high temporal frequencies. Multiple push beams were generated with multiple focused beams and multiple intersecting unfocused beams. Measurements made with tonebursts of ultrasound with length 200 μs repeated at 500 and 1000 Hz were compared against a single toneburst. Measurements were made in commercial homogeneous and heterogeneous phantoms. Wave velocity images were reconstructed from the acquired data. The accuracy and variance were evaluated in the homogeneous phantoms. The contrast and accuracy were evaluated in the heterogeneous phantoms. In both types of phantoms, the magnitude at selected frequencies will be compared among the different excitations. The frequency bandwidth for each excitation will be compared. This work highlights that the combination of multiple push beams and excitations with high temporal frequencies can be combined for improved imaging in shear wave elastography.

9:55–10:10 Break

10:10

2aBAa7. Effect of acoustic impedance mismatch between skin and bone on transcranial ultrasound transmission. Shreyank Gupta (Mech. Eng., École de technologie supérieure, 1100 Notre-Dame St. W, Montreal, QC H3C1K3, Canada, shreyank.gupta.1@ens.etsmtl.ca), Guillaume Haïat (CNRS, Laboratoire Modélisation et Simulation Multiéchelle UMR CNRS 8208, Creteil, France), Catherine Laporte (Elec. Eng., École de technologie supérieure, Montreal, QC, Canada), and Pierre Belanger (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

A major problem with transcranial Doppler (TCD) ultrasound is the poor transmission of ultrasound through the skull bone causing image quality degradation. The reasons for the poor image quality are (1) acoustic impedance mismatch along the wave propagation path and (2) bone frequency-dependent attenuation. Transmission loss due to acoustic impedance mismatch is typically ignored in the literature. Therefore, the objective of this paper is to study the effect of acoustic impedance mismatch on the transmitted energy as a function of frequency. To achieve this, the wave propagation was modelled analytically from the ultrasonic transducer into the brain. The model calculates frequency-dependent transmission coefficient for a given skin and bone thickness combination. The model incorporates both attenuation and acoustic impedance mismatch effects. The model was validated experimentally by comparing with measurements using a bone phantom plate mimicking the acoustic properties of the skull. The average error on the skin thickness between the model and the experiment was less than 6% for a constant bone thickness. Further analysis of the simulation suggests that an optimized excitation frequency can be chosen based on skin and bone thicknesses that would improve ultrasound transmission.

10:25

2aBAa8. Ultrasonic evaluation of anisotropic structure of swine skull. Nagomi Murashima (Life and Medical Sci., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe-shi, Kyoto 610-0321, Japan, ctub1035@mail4.doshisha.ac.jp), Leslie V. Bustamante Diaz (Sci. and Eng., Doshisha Univ., Kyoto, Japan), and Mami Matsukawa (Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

For the ultrasound treatment in brain, such as high intensity focused ultrasound (HIFU), understanding of the complicated ultrasonic propagation in the skull bone is important, because of the anisotropic and heterogeneous character. In this study, we experimentally investigated the ultrasonic wave propagation in the skull sample, considering the anisotropic structure. A cylindrical bone sample was fabricated along the thickness direction of the right occipital bone of a swine (diameter: 11.5 mm, height: 3.0 mm). The ultrasound velocity in the MHz range was measured in the radial direction by rotating the sample (0 degree: sagittal-axis) and changing the measurement position. The structure of the sample was also evaluated by the X-ray micro CT. There are three orthogonal trabecular alignments in bone sample: main, intermediate-shafts (IS) and minor. In the outer layer of the skull, IS were mostly found along the frontal-axis. Ultrasound velocity became higher as the wave propagation direction approached to IS and values were in the range from 2110 to 2790 m/s. In the dipole, ultrasound velocity became higher as the wave propagation direction approached to main-axis and values were in the range from 2080 to 2600 m/s.

10:40

2aBAa9. Validation of single-element ultrasound targeting in the brain for human neuromodulation. Joseph Blackmore (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, joseph.blackmore@wadh.ox.ac.uk), Verena Braun, Christopher Butler (Nuffield Dept. of Clinical NeuroSci., Univ. of Oxford, Oxford, United Kingdom), and Robin Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Non-invasive brain stimulation techniques allow for excitation or inhibition of neural activity via externally applied stimuli through the skull. Recently, ultrasound has been shown to stimulate neurons with better spatial localization than existing electrical stimulation modalities and has subsequently been employed in several human neuromodulation studies. While increased focality provides the potential for modulation of specific brain targets, the overlying skull bone can distort the ultrasound beam leading to shifting and fragmentation of the focus. Consequently, care must be taken to ensure the ultrasound focus is coincident with the intended brain target to confirm the induced effects are due to direct ultrasound activation. Here, we present numerical simulations using k-Wave and experimental measurements demonstrating focusing to different areas of the visual cortex using ex-vivo human skulls with single-element transducers. We show that targeting of many areas can be achieved without the use of lenses or arrays. The sensitivity of transducer angle and location on the resultant pressure field within the skull is assessed along with comparisons between MRI and CT datasets and elastic and fluid-based simulations. Finally, we also explore correction methods for targeting areas of the brain through more complex skull topologies, for example, through the petrous ridge.

Invited Papers

10:55

2aBAa10. Simulation of ultrasound propagation through human skull: Experimental validation and application to treatment planning. Frederic Padilla (Focused Ultrasound Foundation, 151 Cours Albert Thomas, Lyon 69390, France, frederic.padilla@inserm.fr), Raphaël Loyet (LabTAU INSERM U1032, Lyon, France), David Moore (Focused Ultrasound Foundation, Charlottesville, VA), Achour Ouaked (LabTAU INSERM U1032, Lyon, France), John Snell, Matt Eames (Focused Ultrasound Foundation, Charlottesville, VA), SYLVAIN Chatillon (LIST, CEA, Gif sur yvette, France), and Cyril Lafon (LabTAU INSERM U1032, Lyon, France)

Transcranial focused ultrasound (FUS) is a non-invasive therapeutic modality that can be used to treat essential tremor via local thermal ablation of a small spot in the thalamus. Acoustic energy is focused through the skull using a multi-element transducer. Per-element phase delay is derived using individual patient's skull CT to compensate for skull heterogeneity, but MRI thermometry is still required for precise targeting and localization of the focal spot. There exists an opportunity for improved and accurate numerical simulations through skull to improve focal spot positioning and treatment, to allow treatment pre-planning, and to permit parametric studies that will identify important acoustic and thermal properties of human skull bones and tissues. Here, we report on a novel 3D numeric simulation framework, based on the CIVA Healthcare simulation platform, to simulate propagation through skulls and the resulting heating. This simulation platform has been developed to pilot various fast algorithms and to simulate the heating in tissues induced by FUS by solving Pennes' BioHeat Transfer Equation. Simulations of propagation through human skulls with the Insightec 650 kHz clinical system, with 1024 elements, were performed and validated by comparison to experimental acoustic signals acquired with a hydrophone and to MRI thermometry data.

11:15

2aBAa11. Implementation of a dynamic ray tracing method in CIVA HealthCare HIFU simulation platform—Application to the propagation in inhomogeneous and heterogeneous tissues. Sylvain Chatillon (LIST, CEA, Institut CEA LIST CEA Saclay, Bât. Digiteo - 565, Gif sur yvette 91191, France, sylvain.chatillon@cea.fr), Raphaël Loyet, Françoise Chavrier (LabTau, INSERM, Lyon, France), Ayache Bouakaz (U 930, INSERM, Tours, France), Jean-Yves Chapelon (LabTau, INSERM, Lyon, France), Stéphane Leberre, and Pierre Calmon (LIST, CEA, Gif sur yvette, France)

For several years, CEA-LIST has been developing in partnership with INSERM, a HIFU simulation platform, CIVA Healthcare. It provides specific tools for the development and optimization of probes and therapeutic protocols in order to target clinical problematic of specific organs. This platform allows to easily simulate 3D pressure field induced by HIFU, in linear and non-linear regimes, and the corresponding thermal dose in tissues and phantoms. In this communication, we propose a particular focus on the dynamic ray method for linear acoustic propagation. This model enables fast simulations of HIFU propagation in heterogeneous configurations, including soft and hard tissues, made of isotropic and anisotropic medium, with homogeneous or inhomogeneous elastic properties. The contribution of a source point located on the emitting surface is obtained considering the evolution of a pencil of rays centered on the geometrical path linking this source and the observation point. This models takes into account the propagations of bulk waves in each medium and the various reflections and refractions, with and without mode conversion, at each interface. Examples of computation of pressure fields through hard tissues (bones, skull) and in heated soft tissues will be presented. [Work supported by French Nation Research Agency (ANR SATURN -15-CE19-0016).]

Contributed Paper

11:35

2aBAa12. A ray acoustics-based simulation for predicting trans-vertebral ultrasound propagation: Simulation accuracy. Rui Xu and Meaghan A. O'Reilly (Sunnybrook Res. Inst., 2075 Bayview Ave., Apt. 200, Toronto, ON M4N 3M5, Canada, xurui.xu@mail.utoronto.ca)

The blood-spinal cord barrier (BSCB) prevents drug delivery to spinal cord parenchyma. Focused ultrasound can temporarily increase BSCB permeability in small animal models. The human vertebral column distorts ultrasound foci, preventing clinical translation of this technique. Beamforming must be used to correct focal distortion and may be performed using ray acoustics and patient-specific preoperative CT scans. We evaluate the simulation accuracy of ray acoustics, applied to trans-vertebral ultrasound propagation, through comparison with experiment. Trans-vertebral ultrasound propagation was measured for a spherically focused transducer geometrically focused to the centre of individual thoracic vertebral foramen. Simulation pressures were compared to experiment pressures. Simulation error in voxel pressure was evaluated using root-mean-square error, and was similar to error in a water-only case. Average simulation error across all measurements and simulations in maximum pressure location and weighted >50% focal volume location were 2.3 mm and 1.5 mm, respectively. Simulation error is small relative to the dimensions of the transducer focus (4.9 mm full width half maximum), the spinal cord (8–10 mm diameter), and vertebral canal diameter (15–20 mm diameter), suggesting that ray acoustics may be sufficiently accurate for ultrasound beamforming to the vertebral foramen.

TUESDAY MORNING, 6 NOVEMBER 2018 SIDNEY (VCC), 9:15 A.M. TO 11:55 A.M.

Session 2aBAb

Biomedical Acoustics and Physical Acoustics: Targeted Drug Delivery–Acoustic Radiation Force

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii -Manoa, 2540 Dole Street, Holmes Hall 302, Honolulu, HI 96822

Alfred C. Yu, Cochair
University of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada

Chair's Introduction—9:15

Invited Papers

9:20

2aBAb1. Time course of the vasoactive response for sonoreperfusion therapy using long tone burst ultrasound. François Yu (Radiology, Université de Montréal, 900 rue Saint Denis, R11-422, Montréal, QC H2X0A9, Canada, francois.yu@umontreal.ca), Xucai Chen, Jianhui Zhu, Flordeliza Villanueva, and John Pacella (Vascular Medicine Inst., Univ. of Pittsburgh, Pittsburgh, PA)

We have previously shown that sonoreperfusion **(SRP)** using ultrasound **(US)** and microbubble **(MB)** can restore perfusion following microvascular obstruction and is mediated by nitric oxide vasodilation. Here, we compared the use of short (**SP**-10 cycles) and long (**LP**-5000 cycles) pulses for SRP and investigated the vasoactive response time-course and the downstream signaling 4 h post-treatment. In a rat hindlimb, we applied LP or SP (1 MHz,1.5 MPa) with the same number of acoustic cycles for 2 min during MB infusion. Both pulses affected the perfusion rate but with distinct temporal responses. For SP, the flow rate peaked (12.3±3.4 dB/s) at 15 min but returned to baseline **(BL)**(3.2±0.8 dB/s) after 60 min. Conversely, for LP, the flow rate also initially dropped but started to increase at 6 min and peaked (13.0±2.1 dB/s) at 15 min and was still high (8.3±2.0 dB/s) after 4 h compared to BL and SP. This sustained increase in flow rate, which was different from reactive hyperemia, was concordant with 2.5-fold increases in phosphorylated and total eNOS expression (Western blot) compared to untreated tissue, and consistent with a 2-fold increase in eNOS transcription (qPCR). These data support that long pulse SRP therapy increased in muscle perfusion for at least 4 h consistently with increased eNOS transcription and expression.

9:40

2aBAb2. Feasibility of using ultrasound with microbubbles to purify cell lines for immunotherapy application. Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington; Phys. Faculty, Moscow State Univ., Seattle, Washington), Lev Ostrovsky (Dept. of Appl. Mathematics, University of Colorado; Inst. of Appl. Phys., Russian Acad. of Sci., Boulder, CO), Andrew Brayman, John Kucewicz, Brian MacConaghy, and Dino De Raad (Univ. of Washington, Seattle, WA)

Cell-based immunotherapies exploit cell surface antigens to identify and purify cell lines. Fluorescence-based sorters require large sample volumes and are too costly for small labs. Magnetic sorters require enzymatic digestion to remove magnetic particles. We propose to label cells with antibody-conjugated microbubbles (MBs) and selectively sort them using ultrasound. After sorting, the MBs can be removed by a small overpressure. TargeStar-SA microbubbles (MBs) were conjugated to leukemia cells expressing CD7 antigens. Conjugated cell suspensions were placed in a flow with erythrocytes (which lack CD7) and imaged under magnification. Cell motion was quantified with or without ultrasound insonation. The acoustic radiation force (ARF) acting on a cell-MB pair was modeled by assuming the driving force is associated with the MB, and the viscous drag is due to the larger cell. A separate observation of cell-MB rotation caused by the ARF was explained by considering the torque on the cell, where adherent MBs act as point forces. Under insonation, tagged cells were observed to rotate to align with the ARF, and to move in the direction of the ARF. If a tagged cell was adherent to the coverslip, it only rotated. The model for rotation-only behavior fit well with the data. Under flow, tagged leukemia cells were deflected by ultrasound, while erythrocytes were unaffected. These initial studies suggest a new way for isolating and sorting cells. [Funded by LSDF #3292512, RBBR 17-02-00261, NIH P01 DK43881, and NSBRI via NASA NCC 9-58.]

10:00

2aBAb3. Cytomechanical perturbations induced by pulsed ultrasound and acoustic cavitation. Alfred C. Yu (Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca) and Jennifer M. Wan (Univ. of Hong Kong, Pokfulam, Hong Kong)

The therapeutic applicability of ultrasound is perhaps well demonstrated with the advent of high-intensity focused ultrasound (HIFU) that works by rapidly heating tissues to induce ablation. Ultrasound also holds tremendous therapeutic potential at low intensities that are near or below the clinical diagnosis limit (720 mW/cm^2 spatial-peak time-averaged intensity, as set forth by FDA), presumably acting through a mechanical interaction pathway. To establish the therapeutic potential of low-intensity ultrasound, it is fundamentally important to characterize its biophysical interactions with living cells. This talk shall present a series of non-thermal wave-matter interactions of low-intensity pulsed ultrasound. The biophysical interactions between low-intensity ultrasound and living cells will particularly be demonstrated through direct observations acquired using live optical and confocal microscopy tools. We will also examine the scenario where microbubbles are introduced as agents to induce acoustic cavitation. Temporary membrane perforation may be readily achieved in this case (often referred to as sonoporation), and it has been tipped as an emerging paradigm for drug/gene delivery. Epitomizing observations on this process will be shown during the presentation.

10:20–10:35 Break

10:35

2aBAb4. Acoustic fields and forces in drug delivery applications. John S. Allen (Mech. Eng., University of Hawaii -Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

For ultrasound drug delivery applications, the ability to manipulate and move both the carriers and their payloads provides a way of increasing over efficacy. Both primary and secondary acoustic radiation forces have been used though the optimal acoustic forcing parameters and delivery system are subject to on-going investigations. The primary radiation or Bjerknes force occurs from inhomogeneous propagation of the acoustic field and has been used to direct targeted ultrasound contrast agents or particles in vessels towards endothelial cells or pathological targets at the vessel wall. Secondary Bjerknes forces arise from multiple scattering effects between neighboring particles. For corresponding attractive force regimes, greater particle congregation is possible. Overview of theoretical background and predictions of the acoustic radiation forces is given. The formulation of particle translation by primary radiation in the long wavelength limit is discussed with respect to unsteady drag. Pair wise interaction between particles moving in tandem is compared with formulations for along the lines of center. Nonstationary forcing with respect to amplitude modulation or frequency (chirp) alters the attraction and repulsion regimes compared to continuous forcing. Cellular transport for payloads may be enhanced using more optimally tuned pulse sequences.

10:55

2aBAb5. Cloaked microbubbles and acoustic radiation force for ultrasound molecular imaging. Mark Borden (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., Campus box 427, Boulder, CO 80309, mark.borden@colorado.edu), Shashank Sirsi (Mech. Eng., Univ. of Colorado, Dallas, TX), Jason Streeter, and Paul Dayton (Biomedical Eng., UNC/NCSU, Chapel Hill, NC)

In designing targeted contrast agent materials for imaging, the need to present a targeting ligand for recognition and binding by the biomarker is counterbalanced by the need to minimize interactions with plasma components, and to avoid recognition by the immune system. We have designed a microbubble imaging probe for ultrasound molecular imaging that uses a cloaked surface architecture to minimize unwanted interactions and immunogenicity. Here, we examine the utility of this approach for *in vivo* molecular imaging. In accordance with previous results, we showed a threefold increase in circulation persistence through the tumor of a fibrosarcoma model in comparison to controls. The cloaked microbubbles were then activated for targeted adhesion through the application of the primary acoustic radiation force applied specifically to the tumor region. Using a clinical ultrasound scanner, microbubbles were uncloaked, imaged and silenced. Results showed a twofold increase in ultrasound radiation force enhancement of acoustic contrast intensity for cloaked microbubbles, while no such increase was found for exposed-ligand microbubbles. We conclude that use of cloaked microbubbles for ultrasound molecular imaging bridges the demand for low immunogenicity with the necessity of maintaining targeting efficacy and imaging conspicuity *in vivo*.

11:15

2aBAb6. Shape stability of an encapsulated microbubble undergoing translational motion. Yunqiao Liu (Key Lab. of HydroDynam., Shanghai Jiao Tong Univ., Shanghai, China), Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu), and Qianxi Wang (School of Mathematics, Univ. of Birmingham, Birmingham, United Kingdom)

Encapsulated microbubbles (EMBs) are associated with a variety of important diagnostic and therapeutic medical applications, including sonography, drug delivery, and sonoporation. The nonspherical oscillations, or shape modes, of EMBs strongly affect their stability and acoustic signature, and thus are an important factor in the design and utilization of EMBs. Under acoustic forcing, EMBs often translate with significant velocity, which can excite shape modes, yet few studies have addressed the effect of translation on the shape stability of EMBs. To investigate this phenomenon, an axisymmetric model is developed for the case of small shape oscillations. The exterior fluid is modeled as potential flow using an asymptotic analysis. Viscous effects within the thin boundary layer at the interface are included, owing to the no-slip boundary condition. In-plane stress and bending moment due to the encapsulation are incorporated into the model through the dynamic boundary condition at the interface. The evolution equations for radial oscillation, translation, and shape oscillation of an EMB are derived. These equations are solved numerically to analyze the shape mode stability of an EMB and a gas bubble subject to an acoustic, traveling plane wave. The findings demonstrate the counterintuitive result that translation more readily destabilizes an EMB than an uncoated gas bubble. The main factor responsible for mediating this translation-induced interfacial instability is the no-slip condition at the encapsulating membrane.

11:35

2aBAb7. Insights into the roles of radiation forces in bubble mediated sonothrombolysis. Christopher Acconcia (Univ. of Toronto, Toronto, ON, Canada), Alex Wright (Sunnybrook Res. Inst., Toronto, ON, Canada), Kevin Kiezun, and David Goertz (Univ. of Toronto, S665a, 2075 Bayview Ave., Toronto, ON m4k1x5, Canada, goertz@sri.utoronto.ca)

It is increasingly recognized that radiation forces on bubbles are relevant to a range of therapeutic applications such as sonothrombolysis. In this paper, we discuss work using high speed microscopy to investigate the interactions between ultrasound stimulated bubbles and the boundary of biologically relevant materials. Initial studies with fibrin gels (clots), demonstrated that bubbles could be stimulated to deform, penetrate, and damage the fibrin network. This highlighted two distinct processes where radiation forces play a central role: transporting bubbles from a vessel lumen to a target boundary and facilitating therapeutic effects at the target. To investigate the transport process, experimental data were acquired to assess translational bubble dynamics as a function of size and exposure condition, and this data were used to calibrate a model where both shell and history forces are captured. Experiments and modeling showed that both exposure scheme and flow rate can profoundly influence the number and size distribution of bubbles reaching a target site on a vessel boundary. Once at the boundary, cyclical deformations are induced by the bubble extending well into the material, and these are asymmetric on the forcing and recovery cycles. Deformation, as well as penetration and damage are found to be highly dependent on bubble size and exposure schemes. An improved understanding of these complex processes may facilitate improvements in applications involving large vessels, such as sonothrombolysis.

TUESDAY MORNING, 6 NOVEMBER 2018 CRYSTAL BALLROOM (FE), 9:00 A.M. TO 11:55 A.M.

Session 2aMU

Musical Acoustics and Signal Processing in Acoustics: Modeling Musical Instruments and Effects

Scott H. Hawley, Cochair
Chemistry & Physics, Belmont University, 1900 Belmont Blvd., Nashville, TN 37212

Vasileios Chatziioannou, Cochair
Department of Music Acoustics, University of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Building M, Vienna 1030, Austria

Chair's Introduction—9:00

Invited Papers

9:05

2aMU1. String synthesis using individually modeled termination scattering filters. Mark Rau, Julius O. Smith, and Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 1618 Sand Hill Rd., Apt. 414, Palo Alto, CA 94304, mrau@ccrma.stanford.edu)

In the context of digital waveguide synthesis of string instruments, strings are modeled as pairs of digital waveguides which encounter a scattering junction at each boundary. The string scattering junction at the bridge can be modeled to match the measured driving point admittance at the bridge of a physical instrument, and is used in calculating the radiated sound. The string termination at the nut, or fingering end of the string, is typically modeled with a single scalar attenuation, independent of the fingering. However, driving point admittance measurements are presented confirming that string reflectance is a function of both frequency and fingering. A model is proposed which uses a unique, frequency-dependent string reflection scattering junction for each fingering position. The fingering location scattering junctions are calculated using a modal architecture based on driving point admittance measurements made on an acoustic guitar at the bridge, nut, and frets 1 through 13. These synthesized strings are compared to recordings of the physical instrument, as well as to the widely used scalar attenuation model.

9:25

2aMU2. Power-preserving nonlinear modal coupling, feedback frequency/phase modulation, and the stretched allpass filter. Tamara Smyth and Jennifer S. Hsu (Music, Univ. of California San Diego, 9500 Gilman Dr. MC 0099, La Jolla, CA 92093-0099, trsmyth@ucsd.edu)

Motivated by the sound produced by nonlinear modal coupling in acoustic systems, e.g., changes of sounding frequency in the form of pitch glides/chirps and the generation of harmonics, we model a nonlinear coupling between two modes of oscillation and derive expressions mapping "musical" parameters to the model's "synthesis" parameters. We begin by showing that a unitary power-preserving matrix formulation of the coupled modes leads to a feedback frequency (FM) equation, the instantaneous frequency of which is integrated over time to yield the phase in the corresponding—and preferred—phase modulation (PM) representation. Though the integration is made difficult by the feedback term, an analytic solution is presented by first observing that the real/imaginary parts of the PM equation are equivalent to the real/imaginary parts of the transfer function for a "stretched" allpass filter—one where the delay is not constrained to be a single sample and where the real part produces a periodic comb-like signal, that if made to vary over time (rather than frequency), produces the sounding frequency (a "musical" parameter) resulting from the nonlinear coupling. Ultimately, it is hoped this work may provide computationally efficient models of percussion instruments with rich nonlinear behaviour at little additional computational cost.

9:45

2aMU3. Efficient rendering of saxophone sound by modal synthesis and wave scattering. Esteban Maestre (CAML, McGill Univ., Barcelona, Spain), Gary Scavone (CAML, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, gary@music.mcgill.ca), and Julius O. Smith (CCRMA, Stanford Univ., Stanford, CA)

We present a digital filter design technique for the modeling of impedance and radiation transfer functions as measured from an alto saxophone. For each fingering position, both the input impedance and sound radiation are jointly modeled via a recursive parallel filter structure akin to a modal expansion, with the filter coefficients estimated in the frequency domain by means of constrained pole optimization and least-squares. For modeling the transition between fingering positions, we propose a simple model based on linear interpolation of impedance and radiation models. For efficient sound rendering, the common impedance-radiation model is used to construct a joint reflectance-radiation filter realized as a digital waveguide termination that is interfaced to a reed model based on non-linear scattering.

Contributed Papers

10:05

2aMU4. Physics-based resynthesis of clarinet performance using numerical optimization. Vasileios Chatziioannou, Montserrat Pàmies-Vilà, Sebastian Schmutzhard, and Alex Hofmann (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)

The functioning of musical instruments has been an active area of research since the beginning of the last century, using both theoretical approaches and experimental measurements. Regarding wind instruments, the subtleties of embouchure control are of particular interest, since the actions of the player's lips and tongue evade direct measurement. One approach towards analyzing such actions is by extracting relevant information from signals that allow non-intrusive measurements, such as the blowing pressure, the mouthpiece pressure and the reed bending [Pàmies-Vilà *et al.* (2018) Frontiers in Psychology 9: 617]. Alternatively, it is possible to formulate a physical model of the player-instrument interaction and estimate the actions of the player while resynthesizing the oscillations of the instrument [Chatziioannou *et al.* (2012) Acta Acustica united with Acustica 98(4): 629–639]. This work presents such an attempt based on measurements carried out on expert clarinet players. An excerpt of Weber's clarinet concerto No. 2 is performed by the players and resynthesized using a refined physical model taking embouchure effects into account. Using inverse modeling, the underlying model parameters are being tracked in order to extract information on the physical phenomena that take place at the excitation mechanism.

10:20–10:40 Break

10:40

2aMU5. Acoustical impedance of brass instruments: Measurements and modeling for undergraduate physics course. Kurt Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Whitman College, Walla Walla, WA 99362-2067, hoffman@whitman.edu)

We will present recent work developing experimental and modeling exercises for an intermediate physics course focusing on vibrations and waves with applications to acoustics. The measurements described here utilize a simple acoustical impedance measuring system that can be used

to identify the resonant frequencies of a variety of brass instruments. For modeling purposes, we will have the students build approximate brass instruments from PVC tubing and couplers to simplify the shape of the instrument. Despite the abrupt changes in radius of the PVC instrument, adjustment of the cylindrical tube lengths results in a regular pattern of resonances that matches quite well to the first five or six harmonic intervals required in brass instruments. We have developed a Mathematica based program to model the impedance of the PVC brass instruments. The model results can be compared to the measured impedance resonances of the tube. In addition, the resonances can also be measured by lip buzzing the instrument to demonstrate the modification of the resonance frequencies by the vibrating lips at the mouthpiece. We will also discuss the some pedagogical benefits of using acoustical systems as an introduction to solving differential equations and boundary value problems.

10:55

2aMU6. Profiling musical audio processing effects with deep neural networks. Scott H. Hawley, Benjamin L. Colburn (Chemistry & Phys., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu), and Stylianos I. Mimilakis (Fraunhofer Inst. for Digital Media Technol., Ilmenau, Germany)

Deep learning has demonstrated great performance in audio signal processing tasks such as source separation, dereverberation, and synthesis. A challenging effort in deep learning is to devise models that operate directly on the raw waveform signals (i.e., end-to-end). In this talk, we present attempts of an end-to-end task in audio signal processing, that we denote as profiling (i.e., emulation). Our objective is deep learning based profiling of a set of audio production and editing effects, including nonlinear time-dependent effects such as dynamic range compression and time alignment, as well as learning their parameterized controls (e.g., gain, attack time). We present some promising initial results. A consequence of using a data-driven approach to effects modeling is that it allows for the creation of novel audio effects purely via the construction and training on appropriate datasets, without explicitly devising a signal processing algorithm.

11:10

2aMU7. Dynamic characterization of grand and upright piano action. Philip P. Faraci, John A. Case, Jillian M. Farrell, Eoin A. King, and Robert Celmer (Acoust. Program and Lab., Mech. Eng. Dept., Univ. of Hartford, 200 Bloomfied Ave., West Hartford, CT 06117, faraci@hartford.edu)

A dynamic characterization was performed on a grand piano and an upright (vertical) piano action assembly. The action assembly consists of a key and a system of levers that transfers a force from the key to the hammer. Measurements of the hammer acceleration profile for a given key force were compared to dynamic simulation results from a computer model constructed in SolidWorks. Single-key action assemblies typically used by piano technicians were used for the dynamic measurements and as a guide for the physical dimensions of the computer model. Measurements of hammer velocity were made using a Laser Doppler Vibrometer, which was differentiated to determine the hammer acceleration. The purpose of the study was to establish a relationship between key force and hammer acceleration for the action assemblies and to construct a computer model whose dynamic behavior closely matches the measured profile. Preliminary results between the measured hammer acceleration and the simulated acceleration will be discussed. [Work supported by a University of Hartford CETA Student Engagement Grant.]

11:25

2aMU8. Characterization and modeling of the Erhu. Chris Waltham and Laura Kim (Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada, cew@phas.ubc.ca)

The erhu is a two-string bowed instrument characterized in Chinese organology as a silk instrument (owing to the traditional string material), and by Hornbostel and Sachs as a spike fiddle. The overall length of the instrument is about 80 cm, and the vibrational length of the strings is around half of that. The soundbox is a tube (hexagonal, octagonal, or cylindrical) of about 13 cm length and 9 cm diameter. One end of the tube is covered by a membrane of tensioned python skin and the other is mostly open, usually with a grille. The two strings are stretched over a small bridge positioned centrally on the membrane. In spite of the great difference in soundbox structure between the erhu and a Western violin, the radiated sounds of the two instruments have a lot in common with each other (and with the human voice). Vibrational and acoustic measurements have been made on several erhus: cavity modes, surface velocities and radiated sound. A straightforward coupled-oscillator model has been found to reproduce many of the measurements, and also to allow mechanical characterization of the membrane *in situ*.

11:40

2aMU9. Heuristic finite element models of the Japanese *Koto*. Iran Sanadzadeh and Angela K. Coaldrake (Elder Conservatorium of Music, Univ. of Adelaide, Rm. 8.08, Schulz Bldg., 17 Kintore Ave., Adelaide, SA 5005, Australia, iran.sanadzadeh@adelaide.edu.au)

Modeling a traditional, hand-carved performance-grade Japanese *koto* presents many challenges. A COMSOL Multiphysics finite element model of the instrument, based on CAT scan data, is the most rigorously accurate solution to date. However, a substantially less computationally intensive idealized box model, for example, can provide qualitative assessment of its performance. This paper discusses a COMSOL model based on the physical properties of a solid plank of Australian *paulownia* wood, its validation, and its elaboration into an idealized but unstable hollow box with internal struts and sound holes of varying geometries. It then reports on the development of the next stage model which has 12 cross-sections lofted along a spline to create curvature as the natural progression towards realism. Comparison of these three models and the complex CAT scan model of the *koto* is presented. It shows that the box and lofted models are useful in qualitatively interpreting the results of the complex model. Isotropic models, as used in the literature, were unsuccessful in predicting responses, but using best available approximations for anisotropic elastic constants was helpful. Finally, each stage was able to provide timely feedback to inform the development of the next stage model and guide experimentation.

Session 2aNS

Noise, Physical Acoustics, Structural Acoustics and Vibration, and Architectural Acoustics: Emerging Technologies for Noise Control

Kirill Horoshenkov, Cochair
University of Sheffield, Mappin Street, Sheffield S1 3JD, United Kingdom

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

Chair's Introduction—8:00

Invited Papers

8:05

2aNS1. Future trends in noise control technology. J. S. Bolton (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S. Russell St., West Lafayette, IN 47907-2099, bolton@purdue.edu)

Although acoustics plays an important role in noise control, noise control is arguably more difficult than acoustics owing to the constraints that are usually imposed on noise control solutions by manufacturers. That is, ideally, noise control solutions should occupy no space, weigh nothing, cost nothing, not impact the operation of the device being treated, and remain effective for 20 years without maintenance. Hence, identifying effective noise control solutions is a challenging, multidimensional problem. In this presentation, future trends and opportunities in noise control will be highlighted in four categories: targets, measurement techniques, predictive tools, and noise control methods and materials.

8:25

2aNS2. Noise reduction using metamaterials and metasurfaces. Yun Jing (North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

We are surrounded by noise. Planes, trains, cars, crowds, cooling fans — just about everything is a potential source of noise. Traditional noise abatement methods have a number of limitations and they are particularly ineffective for low frequency noise. As we are becoming increasingly aware of the noise pollution issue, there is a pressing need to develop novel materials for more effective noise reduction. In this talk, I will give an overview of recent development on noise reduction using metamaterials or metasurfaces. I will cover structures such as decorated membrane resonators, hybrid resonators, and coiled Helmholtz resonators.

8:45

2aNS3. Correlating dynamical properties of organic aerogel with increased sound insulation of sandwich wallboard systems. Ning Xiang, Mathew A. Whitney (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu), Sadeq Malakooti (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX), Habel G. Churu (Mech. Eng., LeTourneau Univ., Longview, TX), Nicholas Leventis (Chemistry, Missouri Univ. of Sci. and Technol., Rolla, MO), and Hongbing Lu (Mech. Eng., the Univ. of Texas in Dallas, Richardson, TX)

Low thermal conductivity of classical aerogels has long been exploited, while their dynamic properties are yet to be further investigated. This work investigates a new class of acoustic materials based on porous hierarchical three-dimensional assemblies of organic nanoparticles. Primary polymer nanoparticles are formed via chemical reactions in solution and self-assemble to fractal porous secondary particles, which in turn become the fundamental building blocks of larger globular, fibrous or hybrid structures that form the microscopic network of highly porous monolithic materials referred to as polymer-crosslinked (or X-) aerogels. Their low-cost, thin-panel form has found potential applications as constrained damping layers integrated into gypsum wallboards. The standardized chamber-based random-incident tests demonstrate drastically increased sound transmission losses without significantly increasing the board thickness and weight when light-weight X-aerogel panels of less than 1 cm in thickness are integrated into gypsum wallboards in sandwich structure. For better understanding of its excellent effect as constrained damping layers, this paper discusses experimental investigations on broadband dynamic properties of X-aerogels with the intention to correlate their intriguing dynamic properties with increased sound insulations which will have high potentials as emerging building materials for advanced noise and vibration controls and beyond.

9:05

2aNS4. Complex frequency analysis of the scattering matrix for the design of perfect absorbers in reflection and transmission problems. Noé Jiménez (Instituto de Instrumentación para Imagen Molecular (I3M), Consejo Superior de Investigaciones Científicas (CSIC), Edificio 8B, Acceso N, 1[a] Planta, Camino de Vera s/n, Valencia 46022, Spain, nojigon@upv.es), Vicente Romero-Garcìa, Vincent Pagneux, and Jean-Philippe Groby (UMR CNR 6613, Laboratoire d'Acoustique de la Université du Mans, Le Mans cedex 9, France)

The design of deep-subwavelength structures for low frequency sound perfect absorption is challenging. Subwavelength perfect absorption implies increasing of the density of states at low frequency while maintaining impedance matching to the surrounding medium. Therefore, the study of the eigenvalues and eigenvectors of the scattering matrix in the complex frequency plane appears extremely powerful to analyze such systems and derive the critical coupling condition. The latter consists in exactly compensating the leakage of the structure by using the intrinsic losses. Two simple structures, the thicknesses of which are 88 times and 40 times smaller than the perfectly absorbed wavelengths, are presented for reflexion and transmission problems, respectively.

9:25

2aNS5. Are nano-fibers an emerging noise control solution? Kirill Horoshenkov, Alistair Hurrell (Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk), and Mohan Jiao (Univ. of Sheffield, Sheffield, South Yorkshire, United Kingdom)

This paper presents experimental evidence that laboratory measurements of the acoustical and related non-acoustical data is not a trivial task. This paper also shows that the prediction of the acoustical properties of this seemingly simple material is far from easy. First, there is a problem of characterising the non-acoustical properties of nano-porous membrane with standard laboratory methods. Second, there is a high uncertainty in the value of the membrane thickness which can lead to a substantial change in its predicted acoustical and related non-acoustical properties. Thirdly, there is an argument whether or not sound propagation in a nano-porous membrane can actually be treated as that in a classical porous media. The paper attempts to respond to these questions and to present new evidence that we may not know well yet the true acoustical properties of these fibres. Therefore, the key question is *Are nano-fibres still an emerging noise control solution?*

9:45

2aNS6. Are aerogels an emerging noise control solution or it is just beefed up fiber glass? Kirill Horoshenkov and Hasina V. Begum (Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk)

Recently, there has been a number of publications which suggested that aerogels can offer high acoustical performance in terms of their ability to absorb or to insulate against sound. It seems that a bulk of these publications report laboratory experiments on the acoustical absorption or transmission loss performance for samples of particular thickness and lateral dimensions. None of these results appear to be rigorously linked to the aerogel's chemistry or micro-structure. This paper attempts to fill in this gap in information by explaining what these materials actually are, why they work acoustically and what is the underpinning physical phenomenon of aerogel. It is shown that the main reason that these materials are used in applications is that they are fire retardant or their elastic modulus is relatively independent of frequency and that they highly hydrophobic. It is also shown that the absorption performance of fibre glass impregnated by aerogel is generally controlled by the fibre diameter of fibreglass and by the density of the composite structure.

10:05–10:20 Break

Contributed Papers

10:20

2aNS7. Is anything radically new in noise control research? Kirill Horoshenkov (Univ. of Sheffield, Mappin St., Sheffield S1 3JD, United Kingdom, k.horoshenkov@sheffield.ac.uk) and Mark Swift (Armacell, London, United Kingdom)

This paper attempts to review the state of art of noise control research globally and answer a key question: *Has anything been radically new in noise control research lately?* There is anecdotal evidence that no radical noise control solution has emerged from noise control research since the invention of active noise control by Paul Lueg in 1936 (U.S. Patent 2,043,416). Among passive noise control solutions used for sound absorption the market is dominated by mineral wool and fibreglass (43% in 2017, Global Insulation Market Report, 2024). Technical foams used for acoustic absorption cover pretty much of the rest. Similar situation is for common solutions used for noise and vibration insulation. Their acoustical properties and *in-situ* performance can be predicted within experimental and material parameter uncertainties using a commercial package such as Comsol or semi-empirical models such as by Delany and Bazley. There seem no radical solutions has emerged recently to challenge this status quo. Metamaterials are one promising solution for the future, but they remain largely in the realm of academic laboratories or fancy theories. Therefore, another key question is: *What has true impact of noise control research been so far and what fundamental ideas need to emerge to radically improve this situation?*

10:35

2aNS8. Destructive interferences in resonators created using additive manufacturing. Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca)

The broad absorption of low-frequency noise typically requires the utilization of large thicknesses of sound absorbing materials. As this creates space issues, there is a demand for thin broad low-frequency absorbers. This research aims to develop an acoustic transparent panel for low-frequency broad sound absorption. Destructive sound resonance tubes were created for this scope using modern tools for generating complex geometries in systems of a few centimeters of thickness. The capability of digital fabrication to enable design freedom and to support the production of complex geometries is explored in this paper. Destructive interference means that two interfering sound waves that are in counter-phase, cancel each other. Digital fabrication is shown to play an important role in developing customized sound absorbing components to absorber below 500 Hz, and especially well below 200 Hz. A parametric script using Grasshopper through Rhinoceros to generate automatically the geometric parameters of the 3d resonators is described.

Different materials for 3d printing the resonators are compared. Finally, the challenges of additive manufacturing techniques are finally reported.

10:50

2aNS9. A non-negative artificial neural network configuration for noise predictions. Lucas D. Spies, Sterling McBride, Ricardo Burdisso, and Corina Sandu (Mech. Eng., Virginia Tech, 1706 Boxwood Dr., Apt. D, Blacksburg, VA 24060, lucass19@vt.edu)

Artificial Neural Networks (ANN) provide an accurate relationship between user-defined inputs and outputs for complex problems. Their basic units are called neurons. The computation that takes place in each neuron is a weighted summation of its inputs. These results are then implemented as an input to an activation or transfer function, which in turn defines the final neuron's output. The transfer function configuration is very important when defining the final purpose of the ANN. There are many standard activation functions like pure linear, tangent sigmoid, logistic sigmoid, among others. In this study, an ANN capable of predicting tire pavement interaction noise (TPIN) is constructed. Its outputs correspond to sound pressure in narrowband. Thus, the main requirement is that the outputs must be positive. Several neuron transfer functions were investigated for both hidden and output layers. Finally, a customized hybrid linear-sigmoid transfer function was selected. The final ANN configuration is able to capture typical TPIN spectral content within 400 Hz–1600 Hz. In addition, the error is significantly reduced if compared to those obtained after implementing standard transfer functions.

11:05

2aNS10. Characterization of laptop computer noise and vibration using nearfield acoustic holography. Seonghun Im, Won-Suk Ohm, Inman Jang (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, seonghun.im@yonsei.ac.kr), and Heoungkil Park (Samsung Electro-Mech., Suwon, South Korea)

Noise from a laptop computer can be a source of annoyance, especially when the computer is used in a quiet environment for a long period of time. Apart from the fan noise, an equally troublesome noise, often characterized as the "buzz," is produced by the vibration of the internal electronics components that transmits through the circuit board and eventually manifests itself as sound radiation. Because the buzz is very low in SPL (but loud enough to cause annoyance in a quiet environment), its measurement requires a nearfield technique that is specialized to a low-intensity sound source. In this talk, we discuss the application of nearfield acoustic holography (NAH) for this purpose, and present an acoustic camera system that can measure and visualize the noise and vibration of a laptop computer. The acoustic hologram measured in the nearfield of a laptop computer is utilized to obtain the two-dimensional maps of the acoustic intensity and the velocity of vibration on the surface of the laptop as well as the total radiated power. The NAH-based acoustic camera system described here proves to be an effective tool for characterization and control of noise and vibration present in mobile electronic devices.

11:20

2aNS11. Noise exposure assessment of a modular construction manufacturing factory. Sanam Dabirian, Joonhee Lee, and Sanghyeok Han (Dept. of Bldg., Civil, and Environ. Eng., Concordia Univ., EV 6.218, 1515 Rue Sainte-Catherine O, Montréal, H3G 2W1, Montreal, QC H3G 2W1, Canada, sanamdabirian@gmail.com)

The National Institute for Occupational Safety and Health (NIOSH) has laid down specific regulations to prevent harmful impacts of noise on workers. However, due to the lack of awareness of irreparable noise damages, the potential of the auditory and non-auditory diseases is increasing. In the modular construction manufacturing, which is increasingly applied as a construction method, construction workers are substantially exposed to high-noise levels. In order to provide a healthier workplace, assessment of the noise exposure is necessary while in this industry has not been fully studied yet. Therefore, this paper presents noise exposure assessment of a modular construction factory. After investigating patterns of the tasks, task-based method (TBM) was applied as a measurement strategy. K-means clustering was used to determine the noise level of each task to calculate the noise exposure levels in different locations of the factory. The results of this paper can provide a useful guideline to manage the noise efficiently in the modular construction sector.

11:35–11:50 Panel Discussion

Session 2aPA

Physical Acoustics and Engineering Acoustics: Novel Approaches to Acoustic and Elastic Wave Experimentation: Concepts, Hardware, and Novel Processing Methods

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Dirk-Jan van Manen, Cochair
Geophysics, ETH Zurich, Sonneggstrasse 5, Zürich 8092, Switzerland

Nele Börsing, Cochair
Earth Sciences, ETH Zürich, Sonneggstrasse 5, No. H32, Zürich 8092, Switzerland

Theodor S. Becker, Cochair
Earth Sciences, ETH Zurich, Sonneggstrasse 5, Institute of Geophysics, No. H 41.1, Zürich 8092, Switzerland

Invited Papers

8:15

2aPA1. Time reversal focusing for nondestructive evaluation of cracks. Brian E. Anderson, Sarah M. Young (Dept. of Phys. & Astron., Brigham Young Univ., N145 ESC, N283 ESC, Provo, UT 84602, bea@byu.edu), Marcel Remillieux (EES-17, Geophys. Group, Los Alamos National Lab., Los Alamos, NM), Pierre-yves Le Bas, and Timothy J. Ulrich (Q-6, Detonator Technol., Los Alamos National Lab., Los Alamos, NM)

Time reversal (TR) is a technique that may be used for intentional focusing of wave energy. TR has been developed for crack and defect detection in solid media. Because TR provides a localized focus of elastic energy, it can be used to study the local nonlinear properties at point(s) of interest that are indicative of the presence of cracks and defects. Source transducers may be bonded to the sample under test or to a chaotic cavity that is coupled to a sample under test. A receiver, such as a laser Doppler vibrometer, is placed at a location of interest. Impulse responses are obtained between each source and the receiver and then the impulse responses are reversed in time. The reversed impulse responses are then simultaneously broadcast from the respective source transducers and a focus of energy occurs at the receiver position. Since the largest amplitude is obtained at the receiver position, any amplitude-dependent (nonlinear) vibration that occurs at the receiver position dominates nonlinearities, such as harmonics, that are detected in the signal recorded at the receiver position during focusing. This presentation will review the progress made in the detection of cracks using time reversal and present some recent results.

8:45

2aPA2. Spherical harmonic formulations for passive and active control of acoustic scattering around the sphere. Mihai Orita, Stephen Elliott, and Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd, Southampton, Hampshire SO17 1BJ, United Kingdom, mo4g11@soton.ac.uk)

Acoustic cloaking can be implemented using different strategies: passive, active, or a combination of both. The literature on this topic is rich when it comes to the passive methods, specifically on metamaterial designs. However, active and hybrid methods are less well explored. In this work, spherical harmonic decomposition is used to calculate the scattering from a spherical obstacle subjected to an incident, monochromatic plane-wave. Its surface is defined through boundary conditions equivalent to a uniform, locally reacting impedance (Robin condition). Secondary sources represented by point-monopoles on the sphere are used to form an active control system. In the absence of active control, at low-frequencies, the least scattering is achieved if the surface impedance is negative, purely imaginary and has a large absolute value. At high-frequencies, the best performance is realised with a real, positive impedance that is close to that of the exterior propagation medium. When adding active control, only the low-frequency regime shows improvement, given any impedance, and it is most beneficial to enhance for the previous case that exhibited the best passive performance. Thus, a promising hybrid cloaking approach is a combination of: a stiffened-controlled boundary with multi-channel active control at low-frequencies, and a passive, absorptive boundary at high-frequencies.

9:15

2aPA3. Immersive experimentation: A new paradigm for low-frequency acoustic and elastic wave propagation experimentation.
Dirk-Jan van Manen, Theodor S. Becker, Nele Börsing (Geophys., ETH Zurich, Sonneggstrasse 5, Zürich 8092, Switzerland, dirkjan.vanmanen@erdw.ethz.ch), Henrik R. Thomsen, Miguel Moleron (Geophys., ETH Zurich, Zurich, Switzerland), and Johan O. Robertsson (Geophys., ETH Zurich, Zürich, Switzerland)

Immersive experimentation is a new paradigm for wave-based laboratory experimentation aimed at overcoming the laboratory- and sample-size related limitations of conventional approaches and thereby significantly extending downwards the usable frequency range. Using so-called immersive boundary conditions, a physical experimentation domain or elastic rock volume can be immersed in an arbitrary larger numerical domain in such a way that the waves in the physical domain drive the simulation in the numerical domain and vice-versa. Waves propagating towards the edges of the experimentation domain are sensed by dense receiver arrays and extrapolated to/ decomposed at the boundary where they are actively suppressed. Waves incident on the boundary are also extrapolated through the larger numerical domain using pre-computed Green's functions and re-emitted into the experimentation domain, enabling arbitrary order scattering interactions between the laboratory or sample and the numerical domain. We present two laboratories currently under construction at ETH Zurich: WaveLab, an acoustic lab, which implements immersive experimentation in real-time using a low-latency, high-performance acquisition, compute and control system and matrix, an elastic experiment, which implements immersive experimentation using a robotized three-component scanning LDV and novel methods for wavefield separation and injection at an elastic free surface. Implementation, results, and applications are discussed.

Contributed Papers

9:45

2aPA4. Characterisation of complex fluid media by an angle-scanning ultrasound diffractometer.
Mathew J Francis, Melvin J. Holmes, Malcolm J. Povey (School of Food Sci. and Nutrition, Univ. of Leeds, Leeds, West Yorkshire, United Kingdom), Valerie J. Pinfield (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough University, Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk), Artur L. Gower (School of Mathematics, Univ. of Manchester, Manchester, United Kingdom), Jinrui Huang, Derek M. Forrester (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough, Leicestershire, United Kingdom), William J. Parnell, and Ian D. Abrahams (School of Mathematics, Univ. of Manchester, Manchester, United Kingdom)

The characterisation of complex fluid media (particles in liquids) by ultrasound can provide information on the particle size distribution, concentration, and aggregation of the particles. In the Rayleigh scattering regime, systems of liquid particles are dominated by monopole scatter (potentially producing thermal waves), whereas solid particle suspensions are dominated by dipole scatter (producing shear waves); thus the angular dependence of the diffracted field is expected to be different in these cases. Since mode conversion mechanisms are affected by the proximity of particles and thus by the degree of aggregation, the effect of particle type and degree of aggregation is expected to be observed in the angle-dependent signals. We report an experimental investigation of the angle- and frequency-dependent diffracted signal from cylindrical samples (in tubing) of emulsions and suspensions. The experimental results are compared with theoretical predictions using effective homogeneous properties for the complex fluids derived from multiple scattering models. We identify the differences in the diffracted signals for complex media in which the scattering mechanisms are monopole or dipole-dominated.

10:00

2aPA5. Numerical simulations of acoustic cloaking in a real laboratory that deploys immersive boundary conditions.
Nele Börsing, Theodor S. Becker (Geophys., ETH Zürich, Sonneggstrasse 5, NO H32, Zürich 8092, Switzerland, nele.boersing@erdw.ethz.ch), Andrew Curtis (Mathematical GeoSci., Univ. of Edinburgh, Edinburgh, United Kingdom), Dirk-Jan van Manen (Geophys., ETH Zürich, Zürich, Switzerland), Thomas Haag, Christoph Bärlocher (Geophys., ETH Zürich, Zurich, Switzerland), and Johan O. Robertsson (Geophys., ETH Zürich, Zürich, Switzerland)

Immersive boundary conditions (IBCs) act as a wavefield injection method that couples a physical wave propagation experiment to an arbitrary virtual domain. They allow novel, real-time applications for acoustic wave experimentation such as interactions between physical and virtual domains, broadband cloaking, and holography. The implementation of IBCs relies on actively injecting a particular wavefield on the boundary of the physical domain from a dense array of transducers. The injected wavefield must honour the real-world physical and the virtual or computational domains. To calculate the required inputs, a dense array of recording transducers inside the boundary is used from which the future wavefield arriving at the source transducers can be predicted using a discretised Kirchhoff-Helmholtz integral. Whereas IBCs are effectively exact for purely numerical applications, a physical implementation in a laboratory suffers from limitations mainly associated with spatial and temporal discretization issues and the imprint of the electrical devices such as transfer functions or radiation characteristics of transducers. We present a comprehensive numerical sensitivity study of a cloaking experiment in a 2D acoustic waveguide. This defines physical limitations of real-world IBCs, and systematic errors introduced due to subsampling of the recording and injection surfaces and to radiation characteristics of the transducers.

10:15–10:30 Break

10:30

2aPA6. Study of low-hydrostatic pressure dependent elasticity of porous alumina saturated with various gas media.
Ashoka Karunarathne, Josh R. Gladden (Phys. and Astronomy & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu), and Gautam Priyadarshan (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

Poroelasticity describes how the elasticity of porous materials depend on the material properties: porosity, pore saturation medium and its pressure, nature of pore arrangement, and the skeleton material. The primary goal of this study is to investigate the low-hydrostatic pressure dependent elastic properties of porous alumina saturated with different gas media using resonant ultrasound spectroscopy (RUS). RUS is a precise experimental approach for investigating the elastic properties of solid materials which is capable of measurements at different temperatures and hydrostatic pressures. In RUS, the elastic stiffness tensor of crystalline and polycrystalline solids is determined from the vibrational resonance spectra. In this study, we have used commercially available porous alumina ceramic material which has ~40% of porosity and 2 μm of pore diameter. Here we are reporting the variation of elasticity and acoustic attenuation of porous alumina during the low-hydrostatic pressure cycles from 760 torr (1 atm) to 0.1 torr. An experimentally observed discontinuity of elastic stiffness and acoustic attenuation during the pressure region ~150 torr to 0.1 torr will be described by the transition from viscous to molecular flows, quantified by Knudsen number. The effects of saturated gas medium to the elasticity will also be discussed based on the RUS measurements taken during the air, He, and Ar gas saturation.

10:45

2aPA7. Robust downhole ultrasonic pitch-catch measurements of compressional and shear wave velocities. Naoki Sakiyama (Schlumberger K.K., 2-2-1 Fuchinobe, Chuo, Sagamihara 252-0266, Japan, NSakiyama@slb.com), Evgeniya Deger, Hiroshi Hori, Shin'ichi Houshuyama, Atsushi Oshima, and Hiroshi Nakajima (Schlumberger K.K., Sagamihara, Kanagawa, Japan)

Probing acoustic wave velocities of earth formations, both in azimuth and radius, is important when evaluating anisotropy and/or heterogeneity of the formations. A conventional approach for evaluating these complex formations is to employ a sonic logging tool that operates from approximately 0.1 kHz to 20 kHz with an approximately 600 mm spatial resolution. With these standard sonic tools, evaluating formation acoustic wave velocities with an order smaller spatial resolution to detect thin beds and/or azimuthal variation of formation properties is very challenging. To overcome this challenge, this paper discusses an ultrasonic pitch-catch measurement that can be applied to a downhole acoustic tool. Downhole conditions are typically harsh, especially for ultrasonic frequencies. Thus, a robust measurement system is required that maximizes the signal-to-noise ratio. One of the ways to do this is to minimize the distance between the transmitter and the array receiver (and the receiver-receiver spacings of the array). Numerical modeling corroborated by experimental study indicates such a measurement system can primarily detect refracted compressional waves and surface mode waves related to the pseudo-Rayleigh mode. Characteristics of the pseudo-Rayleigh mode measured with such system and a way to estimate shear wave velocity from the measured mode are discussed.

11:00

2aPA8. Ultrasound elastography unveils the shear wave field: Prospects for medical and geophysical imaging. Johannes Aichele, Chadi Zemzemi, Stefan Catheline (Labtau, INSERM, Cours Albert Thomas 152, Lyon 69003, France, johannes.aichele@inserm.fr), and Philippe Roux (Isterre, CNRS, Grenoble, France)

Transient elastography is an ultrasonic imaging method primarily applied in medical imaging. Through correlation of echography images it reconstructs the particle velocities and displacements of the transversal wave-field inside a medium of interest. Elastography measurements are therefore not limited to an acquisition surface. We present developments in processing and experimental design that broaden the application horizon for elastography imaging. First, we propose a multiple source-array for shear wave beamforming that adaptively recovers the impulse response of the medium through time-reversal. This allows for targeted shear wave focusing and thus significantly enhances the SNR compared to single source elastography. Due to the time-reversal robustness the focusing holds true for transmission through a rigid barrier as well, which we show by applying the method to a human skull model. Second, we present a geophysical laboratory experiment. Combining a friction test-bed with ultrasound elastography enables us to recover the effect of rupture dynamics on shear wave propagation. Albeit the use of hydro-gels our densely sampled wave-field perfectly illuminates typical rupture phenomena. These include super-shear fronts, sub-Rayleigh ruptures, and stick-slip behavior. In the future, passive elastography, an equivalent to seismic coda correlation, could be used for monitoring local elasticity during the rupture.

11:15

2aPA9. Acoustic transducer design for active reflection cancellation in a finite volume wave propagation laboratory. Eli Willard, Michael R. Haberman (Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, eliw@utexas.edu), Dirk-Jan van Manen, Johan O. Robertsson, Theodor S. Becker, and Nele Börsing (ETH Zurich, Zürich, Switzerland)

This report describes the design, fabrication, and experimentally obtained electro-acoustic response of an acoustic transducer suite constructed for use in the Wave Propagation Laboratory (WaveLab) at ETH Zürich. WaveLab aims to immerse a physical acoustic experiment within a real-time numerical environment, by implementing exact boundary conditions (EBCs) [J. Acoust. Soc. Am., **134**(6), EL492–EL498, (2013)]. When scale-model ultrasonic experimentation is not possible, a system with EBCs allows for low frequency, reflection-free acoustic measurements in a small physical domain. The physical experiment of the WaveLab facility consists of a water tank measuring only 2 m on a side, while WaveLab's EBCs are realized through a massive computational engine coupled with a dense array of sensing and emitting acoustic transducers. The transducers are used to sense outgoing acoustic waves and to create a reflection-cancelling surface. Criteria for the transducers are discussed in terms of individual and overall system response. The design parameters and associated models include sensitivity, scattering strength, directivity, frequency response, noise floor, and the dynamic range of the system. The transducer designs and models are presented alongside their physical prototypes and experimental results.

11:30

2aPA10. Physical implementation of immersive boundary conditions in acoustic waveguides. Theodor S. Becker, Nele Börsing, Dirk-Jan van Manen (Earth Sci., ETH Zurich, Sonneggstrasse 5, Inst. of Geophys., No. H 41.1, Zürich 8092, Switzerland, theodor.becker@erdw.ethz.ch), Thomas Haag, Christoph Bärlocher (Earth Sci., ETH Zurich, Zurich, Switzerland), and Johan O. Robertsson (Earth Sci., ETH Zurich, Zürich, Switzerland)

The physical implementation of so-called immersive boundary conditions (IBCs) allows the construction of anechoic chambers, where wavefield reflections from the boundary of a physical domain, such as a wave propagation laboratory, are actively suppressed by emitting a secondary wavefield at the domain boundary that destructively interferes with the reflected waves. Moreover, IBCs enable immersive wave propagation experimentation by linking the wave propagation in the physical domain with the propagation in a numerical domain enclosing the physical domain. In this case, IBCs correctly account for all wavefield interactions between both domains, including higher-order scattering. The physical implementation of IBCs is achieved by densely populating the boundary surrounding the physical domain with transducers that enforce the necessary boundary conditions. The signals required at the injection boundary are predicted with the help of a secondary surface of transducers that record the wavefield on a surface slightly inside the physical domain. The recorded wavefield is extrapolated to the boundary by evaluating a Kirchhoff-Helmholtz integral in real-time using an FPGA-enabled data acquisition, computation and control system. Here, we present results of the active suppression of broadband boundary reflections in 1D and 2D acoustic waveguides.

Session 2aPP

Psychological and Physiological Acoustics, Speech Communication, and Musical Acoustics: Music, Speech, and the Brain

Tian Zhao, Cochair
Institute for Learning and Brain Sciences, University of Washington, Box 367988, Seattle, WA 98195

Patricia K. Kuhl, Cochair
Institute for Learning & Brain Sciences, University of Washington, Box 357920, Seattle, WA 98195

Chair's Introduction—9:00

Invited Papers

9:05

2aPP1. Music, speech, and temporal information processing. Tian Zhao and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu)

Music and speech share many acoustic characteristics. Both utilize dynamic acoustic cues (e.g., frequency, time, intensity) to convey information and emotion. We focus on temporal information processing at a slower time scale (e.g., musical timing, speech rhythm) to examine connections between music and speech: (1) transfer effects, (2) neural mechanisms and (3) theoretical frameworks. I will focus on early infancy. We observed a transfer effect from music experience to both music and speech processing in 9-month-old infants (Zhao & Kuhl, 2016). Infants were randomly assigned to either a 12-session music intervention that focused on triple meter (waltz) or a 12-session control group (no music) during the "sensitive period" for phonetic learning. The music intervention group exhibited enhanced neural detection of both mistimed musical beats and speech syllables, quantified by the mismatch response (MMR) measured with magnetoencephalography (MEG). The effects were observed in auditory and prefrontal regions of the cortex. Following the "predictive coding" and "dynamic attending" theories, I will discuss a current two-pronged approach that can extend this research to understand the effects of music on infants' higher-level cognitive skills as well as their early stage sensory encoding of sounds, and implications of these findings for theories of language development.

9:20

2aPP2. Neural entrainment to missing pulse rhythms. Edward W. Large, Charles S. Wasserman, Erika Skoe, and Heather L. Read (Psychol. Sci., Univ. of Connecticut, 406 Babbidge Rd., Unit 1020, Storrs, CT 06269-1020, edward.large@uconn.edu)

Pulse is the perceptual phenomenon in which an individual perceives a steady beat underlying a complex auditory rhythm, as in music. The neural mechanism by which pulse is computed from a complex rhythm is a topic of current debate. Neural Resonance Theory (NRT) predicts that synchronization itself is the neural mechanism of pulse perception and is supported by studies that demonstrate neural entrainment to complex rhythms. Here we report a behavioral and an EEG study of stimulus rhythms that have no spectral energy at the perceived pulse frequency. A dynamical systems model based on NRT predicts that endogenous oscillation will emerge at the "missing" pulse frequency. We observed 1) strong pulse-frequency steady-state evoked potentials (SS-EPs) to isochronous and missing pulse rhythms, but not to a random control; 2) strong coherence between model-predicted SS-EPs and brain responses for all rhythms; 3) differing pulse-frequency activation topographies for missing pulse rhythms versus isochronous and random controls; and 4) differing frequency-following response (FFR) amplitudes for events in missing pulse rhythms versus isochronous and random rhythms. These observations support the theory that the perception of pulse results from the entrainment of emergent population oscillations to stimulus rhythms.

9:35

2aPP3. Distributed neural systems for musical time processing. Takako Fujioka (CCRMA (Ctr. for Comput. Res. in Music and Acoustics), Dept. of Music, Stanford Univ., 660 Lomita Dri, Stanford, CA 94305-8180, takako@ccrma.stanford.edu)

My talk will feature recent MEG and EEG studies for examining time processing, and, together with other findings, discuss how distributed neural systems for musical time work together in managing top-down and bottom-up prediction, ensemble coordination, and learning-induced neuroplasticity. It has been long known that passive listening of metronome-like isochronous stimuli involves auditory cortices for simple prediction of the stimulus interval and detection of changes. Recently, however, more studies are aimed at capturing natural neural dynamics that encode or predict time without such "perturbation" approach. We show that in addition to auditory cortices, various motor-related brain areas contribute to neural oscillatory power modulations in listening to isochronous stimulus, and that beta-band frequency of this modulation may explain benefits of auditory-sensorimotor training in clinical populations with movement

impairments. Moreover, modulation patterns can encode stimulus intervals, metric hierarchies in perception and imagery, as well as anticipation of upcoming tempo change (e.g., *accelerando* and *ritardando*). Listening paradigm is also used to show neuroplastic changes in naïve subjects after short-term piano training in the reward-related systems. Of further interest is frontal lobe's contribution to musical ensemble coordination sensitive to player's role asymmetry and agency (e.g., human vs. computer).

9:50

2aPP4. Finding the beat: An ethnographic approach to beat perception. Alexander K. Khalil (Inst. for Neural Computation, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0523, akhalil@ucsd.edu)

People commonly use words such as "find," "hold," "keep," and "drop" to describe actions related to the musical beat. While such language implies that musical beats exist in an empirically observable, externalized way, musical beats exist only in our perception. This is most clearly evident when people of different cultures perceive different musical beats while listening to the same musical stimuli. At an early age, the ability to discriminate rhythms uncommon to one's surrounding culture declines steeply. However, it has also been shown that rhythmic training may slow or even reverse this decline. This paper explores this process from an ethnographic perspective, observing the way in which American schoolchildren engage with the different temporal dynamics and aesthetics they encounter as they participate in a multicultural music program across several inner-city schools in Southern California.

10:05–10:20 Break

10:20

2aPP5. Neural convergence of rhythm and language: From trait to state. Leyao Yu (Psych., Vanderbilt Univ., Nashville, TN), Brett Myers, and Reyna L. Gordon (Dept. of Otolarygnology, Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, MCE 10267, Nashville, TN 37232, reyna.gordon@vanderbilt.edu)

As a growing literature has shed light on associations between cognitive mechanisms for music and language, an important role of rhythm and timing is emerging for individual differences in language skills in children. In this talk we will address: A) how musical rhythm relates to individual differences in spoken grammar traits in children ages 5 to 8; and B) potential neural synchronization by which an overlap in processing rhythm and language might occur. Results in the current study show strong correlations between musical rhythm perception and spoken grammar production, converging with prior work. In addition, we found that individual differences in neural synchronization to the speech envelope (measured with EEG) during passive listening are associated with performance on prosodic and musical rhythm behavioral task performance. Additional EEG data (during musical rhythm beat perception) has also been collected from $N = 50$ children with typical language development and $N = 13$ children with grammatical impairments (all children had non-verbal IQ in the normal range); data analysis is underway. In light of these studies and others showing that rhythmic listening can impact grammatical task performance, we will discuss other ongoing translational work that takes first steps towards exploring a causal influence of rhythm on grammar states.

10:35

2aPP6. Redundancy in the speech signal helps amusics perceive prosody. Adam Tierney (Birkbeck College, Birkbeck College, London WC1E 7HX, United Kingdom, a.tierney@bbk.ac.uk)

Speech is information dense, rapidly conveying segmental, semantic, syntactic, and prosodic information through the manipulation of a small handful of acoustic cues. As a result, impaired perception of one of these cues could impede acquisition of linguistic structure at multiple levels. For example, impaired perception of temporal patterns has been linked to developmental language delays. However, auditory impairments do not necessarily lead to language delays, suggesting that some people are able to compensate for difficulties with sound perception. This may be due to the perceptual redundancy of speech—the fact that multiple auditory dimensions often convey the same linguistic information. We tested this hypothesis by examining prosody perception in individuals with amusia, i.e., severe difficulties with the perception of musical melodies. First, we find that amusics show decreased functional connectivity between auditory and motor cortices during speech perception, suggesting that amusia is a domain-general deficit in pitch perception stemming from decreased auditory-motor connectivity. Second, we find that amusics compensate for their impairment by relying upon preserved perceptual abilities, suggesting that redundancy makes speech somewhat robust to individual differences in perceptual skill.

Contributed Papers

10:50

2aPP7. The flow of language: Respiration and the rhythmic motor control of speech. Alexis D. MacIntyre and Sophie K. Scott (Inst. of Cognit. Neurosci., Univ. College London, 17 Queen Square, London WC1N 3AR, United Kingdom, a.macintyre.17@ucl.ac.uk)

Speech rhythm can be described as the temporal patterning by which sequences of vocalic and gestural actions unfold, within and between interlocutors. Despite efforts to quantify and model speech rhythm across languages, it remains a scientifically enigmatic aspect of prosody. For example, the existence and/or form of a basic speech rhythmic unit is hotly contested. One challenge is that the primary means of speech rhythm research has been the analysis of the acoustic signal. As speech is multimodal and motoric, investigations of speech rhythm will likely benefit from a greater range of complementary measures, including physiological recordings. The current experiment explores respiratory effort as a contributing factor in speech rhythm production. Undergoing simultaneous inductive plethysmography and acoustic recording, participants produce speech by reading novel prosaic texts aloud in single and dyadic conditions. Speech-associated breathing patterns are analysed in conjunction with traditional acoustic-phonological approaches to speech rhythm. It is hypothesized that, like spoken utterances, speech breathing is subject to rhythmic constraints. Preliminary results are therefore interpreted with a focus on the temporal relationship between inhalation events and previously proposed phonological rhythmic units (e.g., inter-stress, p-centre, syllable), underscoring breathing as a necessary, yet often overlooked, component in speech rhythm planning and production.

11:05

2aPP8. Improving lyrical intelligibility in live music and concert settings: Evaluating the application of alternative sound reinforcement techniques. Jared H. Koshiol (Integrative Studies, Northern Kentucky Univ., Nunn Dr., Newport, KY 41099, koshiolj1@nku.edu) and Greg DeBlasio (Commun., Northern Kentucky Univ., Highland Heights, KY)

Thirteen individuals were asked to listen to eight short passages of music from two songs containing sung lyrics. They compared the music as it was presented at 94dBA through three different systems; one control emulating a standard live music performance, another with the vocal channel assisted by digital effects (Compression, an Adaptive Limiter, EQ and a Harmonic Exciter), and a third where vocals were removed from the left and right speakers and instead played through a single center speaker of the same make while instruments remained in the left and right speakers. This third design is similar to common mixing of 5.1 theatrical audio. Participants indicated their perception of lyrical intelligibility and pleasurability of the sound for each passage and were asked to ignore artistic choices. Responses indicated that the use of effects processing for vocals somewhat improved lyrical intelligibility and pleasurability for one song, while slightly decreasing these values for the other. The center speaker system for vocals greatly improved (in one case almost doubled) listeners' ability to perceive the lyrics as well as their overall enjoyment of the sound. This improvement is likely linked to spatial hearing; a musical extension of the *Cocktail party effect*.

11:20

2aPP9. Vowel content influences relative pitch perception in vocal melodies: A comparison of models based on brightness vs. intrinsic pitch of vowels. Frank A. Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M4L 3T4, Canada, russo@ryerson.ca) and Dominique Vuvan (Psych., Skidmore College, Saratoga Springs, NY)

Past research involving real and synthesized instrumental timbres has found that note-to-note changes in brightness can influence perception of interval size. Changes that are congruent with changes in pitch led to an expansion, whereas changes that are incongruent lead to a contraction. In the case of singing, the brightness of individual notes (as measured by the spectral centroid) will vary as a function of vowel content. In a recent study, we investigated whether note-to-note changes in the brightness of sung notes were capable of influencing the perception of interval size. While results were consistent with past work on instrumental timbres, we were not able to completely rule out an alternative explanation concerning a perceptual correction for the intrinsic pitch of vowels (e.g., f0 of /i/ tends to be produced higher than /a/). In the present study, we created 288 unique note pairs that varied with regard to absolute change in f0 as well as vowel content. Vowels were sampled from across the vowel space, which allowed us to generate unique predictors for change in brightness (spectral centroid) and changes in intrinsic pitch (F2). Regression analyses will compare the effectiveness of competing models.

11:35

2aPP10. Extended high-frequency hearing enables better talker and singer head orientation detection. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 75 Francis St., Boston, MA 02115, monson@illinois.edu)

Why is the human auditory brain endowed with sensitivity to acoustical energy at frequencies beyond 8 kHz? While this "treble" range is known to be important for music quality, we hypothesize that extended high-frequency hearing capability has been obtained and retained, in part, due to its utility for the detection and perception of conspecific vocalizations (*i.e.*, human speech and voice). More specifically, we hypothesized that access to the highest frequencies enables better detection of a vocalizer's head orientation. To test this hypothesis, we assessed human listeners' ability to detect changes in talker and singer head orientations, and the effect of removing access to the highest frequencies on that ability. Detection of head orientation was significantly impaired by low-pass filtering at 8 and even 10 kHz. One implication is that extended high-frequency hearing loss might impair a listener's ability to detect whether a talker is facing the listener—a skill presumably useful for "cocktail party" listening.

11:50–12:00 Panel Discussion

Session 2aSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustic Metamaterials

Christina J. Naify, Cochair
Jet Propulsion Lab, 4800 Oak Grove Dr., MS 157-316, Pasadena, CA 91101

Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

Contributed Papers

9:00

2aSA1. Numerical investigation of non-reciprocal wave propagation in mechanically-modulated elastic metamaterials. Benjamin M. Goldsberry (Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Acoustic systems that break reciprocity have recently received great attention due to their potential to increase control over wave propagation and to have far-reaching impacts on engineering applications, including more efficient acoustic communication devices and vibration isolation. One means to break reciprocity is by spatiotemporal modulation of material properties, typically via electromagnetic actuation. In the present work, tunable mechanical metamaterials are investigated as a platform to achieve non-reciprocity, where effective dynamic property modulations are induced by an external pre-strain of a two-dimensional cellular material. These elastic metamaterials are studied using a nonlinear finite element model, which is suitable for unit cells having complex geometry and undergoing large deformation caused by the external pre-strain [Goldsberry *et al.*, J. Appl. Phys., **123**, 091711 (2018)]. A small-on-large approximation is used to analyze linear elastic, non-reciprocal wave propagation in the presence of a slowly-varying pre-strain. Results on non-reciprocal wave propagation in negative stiffness honeycombs, a structure exhibiting large stiffness modulations due to the presence of a mechanical instability [Correa *et al.*, Rapid Prototyping J., **21**(2), 193–200, (2015)], are shown as a case example. [Work supported by NSF.]

9:15

2aSA2. Dual-frequency sound-absorbing metasurface based on visco-thermal effects with frequency dependence. Wonju Jeon and Hyeonbin Ryoo (KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, wonju.jeon@kaist.ac.kr)

We propose an acoustic metasurface with near-perfect absorption at two frequencies and design it by using two-dimensional periodic array of four Helmholtz resonators in two types. By considering how fluid viscosity affects acoustic energy dissipation in the narrow necks of the Helmholtz resonators, we obtain effective complex-valued material properties that depend on frequency and on the geometrical parameters of the resonators. We furthermore derive the effective acoustic impedance of the metasurface from the effective material properties and calculate the absorption spectra from the theoretical model, which we compare with the spectra obtained from a finite-element simulation. As a practical application of the theoretical model, we derive empirical formulas for the geometrical parameters of a metasurface that would yield perfect absorption at a given frequency. Whereas previous works on metasurfaces based on Helmholtz resonators aimed to absorb sound at single frequencies, we use optimization to design a metasurface composed of four different Helmholtz resonators to absorb sound at two target frequencies.

9:30

2aSA3. A low frequency underwater meta structure composed by helix metal and viscoelastic damping rubber. Ruihao Zhang (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an Shaanxi, 710072, P.R. China, Xi 710072, China, zhangruihao@mail.nwpu.edu.cn)

Many underwater acoustic absorption materials have been proposed, but good low frequency acoustic absorption remains a challenge. We report an underwater metastructure with excellent acoustic absorption that is constructed by perforating helixes of metal into viscoelastic damping rubber. Finite element analysis shows that this metastructure can achieve an acoustic absorption coefficient of 0.75 at about 100 Hz with a thickness of only 1/68 wavelength at the same frequency. Compared with a homogeneous viscoelastic rubber material, the addition of the helix structure improves the acoustic absorption ability in the range of 0–1,000 Hz. An analysis of vibration displacement maps indicates that waveform transformation, multiple scattering, and reflection energy dissipation mechanisms are the critical factors affecting the absorption performance. Different geometries and materials can adjust the sound absorption characteristics below 1,000 Hz. The proposed metastructure has the advantages of high acoustic absorption ability, broader frequency bandwidth, and regular geometry, providing more possibilities for underwater acoustic manipulation applications.

9:45

2aSA4. Efficient emission of directional sound waves by using subwavelength meta-cavities. Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu), Xudong Fan (Univ. of MS, University, MS), Jiajun Zhao (GOWell Int., LLC, Austin, TX), Bin Liang, Jianchun Cheng (Nanjing Univ., Nanjing, Jiangsu, China), Ying Wu (King Abdullah Univ. of Sci. and Technol., Jeddah, Saudi Arabia), and Maryam Landi (Univ. of MS, Oxford, MS)

Efficient emission of directional sound waves is critical in imaging and communication, yet is held back by the inefficient emission at low frequencies, especially for a small source. A subwavelength enclosure with degenerate Mie resonances was implemented to experimentally enhance the sound power emitted to the far field where the radiation directivity pattern is preserved [L. Maryam *et al.*, Physical Review Letters 120 (11), 114301, 2018]. When considering the efficient emission of directional sound waves, a subwavelength meta-cavity of hybrid resonances can be used to convert the monopole sources to multipole emission [X. Fan *et al.*, Physical Review Applied 9 (3), 034035, 2018] or Mie resonances with spatial asymmetry can be used to even emit directional sound beams [J. Zhao *et al.*, Scientific reports 8 (1), 1018, 2018]. The work offers a practical path toward applications that demand miniaturization of speakers for efficient emission.

10:00

2aSA5. Finite element modeling of fluid-saturated metallic foams from micro-computed tomography. Mark J. Cops, James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, mcops@bu.edu), Elizabeth A. Magliula, and David J. Bamford (Naval Undersea Warfare Ctr. Div., Newport, RI)

Open-cell metallic foams are high stiffness-to-weight cellular materials whose microstructure allows for saturation of viscous liquid. Such a composite has advantages for underwater sound absorption over traditional rubbers due to minimal compression from hydrostatic pressure, composite tunability, and potential for specific gravity less than one. Semi-phenomenological and hybrid numerical models have been shown to predict sound absorption performance of metallic foams, however difficulty arises in determining parameters for the numerical models such as tortuosity, viscous characteristic length, thermal characteristic length, and flow resistivity. Such models also assume that the porous frame is rigid, an assumption valid for only a limited frequency range. In this presentation, finite element models of fluid-saturated metallic foams are created from micro-computed tomography scans and analyzed to determine sound absorption performance. The advantage of this method is that the entire foam microstructure and surrounding fluid can be accurately modeled through a finite element mesh. In addition, experimental measurement of model parameters is not required and the rigid frame assumption can be removed.

10:15–10:15 Break

10:15

2aSA6. An investigation into the direct control of the effective material properties of a fluid medium using active noise control. Joe Tan (ISVR, Univ. of Southampton, Highfield Campus, University Rd., Southampton SO17 1BJ, United Kingdom, j.tan@soton.ac.uk), Jordan Cheer, and Stephen Daley (ISVR, Univ. of Southampton, Southampton, Hampshire, United Kingdom)

Acoustic metamaterials are subwavelength-engineered structures that can exhibit behaviour not seen in conventional materials. The effective material properties for acoustic metamaterials are the bulk modulus and density. Negative effective material properties can create band gaps, where wave propagation is forbidden. Hence, acoustic metamaterials can achieve high levels of noise control performance. Active control has been combined with passive acoustic metamaterials to enhance the level of attenuation or broaden the bandwidth of the band gap. However, the width of the band gaps for these materials are still limited by the use of resonators. This study will investigate whether active control can be employed to directly minimise the effective material properties, which requires an optimization procedure. To begin to understand this optimization problem, the effective material properties have been calculated as a function of the real and imaginary parts of the control source strength at various frequencies for monopole and dipole sources. The resulting error surfaces demonstrate that the two configurations could achieve negative effective material properties; however, the cost functions are not always convex without constraints. Therefore, to adaptively manipulate the effective material properties using an active control system would require the development of an advanced optimization procedure.

Invited Papers

10:30

2aSA7. Acoustic-like behavior in periodic metal structures. Andrew Norris and Xiaoshi Su (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Pentamode materials are of interest for acoustic metamaterials because of their property of having only one elastic mode, similar to the hydrostatic response of an acoustic fluid. This static property does not take into account dynamic frequency dependent effects related to the necessary spatial inhomogeneity in the structural realization. In this talk, we argue that pure pentamode behavior is not desirable, and instead AMM users should focus on the one-wave dynamic response. Experience with metal structures designed to provide acoustic properties close to those of water using metal structures, e.g., Su *et al.* (JASA 2017, doi: 10.1121/1.498519), shows that the most useful response is in a one-wave band gap in which shear waves are non-propagating. Surprisingly, the one-wave region is generally broadband relative to the cutoff frequency of the shear waves at low frequencies, and is non-dispersive, with constant phase speed over the one-wave gap. The talk will attempt to explain these surprising and unexpected features using numerical and analytical results. The latter include an approximation of the dynamic effective properties of 2D and 3D metallic structures displaying broadband one-wave regions using simplistic models.

10:50

2aSA8. Magnetoactive acoustic metamaterials. Qiming Wang (Univ. of Southern California, 920 Downey Way BHE 222, Los Angeles, CA 90004, qimingw@usc.edu)

Acoustic metamaterials with negative constitutive parameters (modulus and/or mass density) have shown great potential in diverse applications ranging from sonic cloaking, abnormal refraction and superlensing, to noise canceling. In conventional acoustic metamaterials, the negative constitutive parameters are engineered via tailored structures with fixed geometries; therefore, the relationships between constitutive parameters and acoustic frequencies are typically fixed to form a 2D phase space once the structures are fabricated. Here, by means of a model system of magnetoactive lattice structures, stimuli-responsive acoustic metamaterials are demonstrated to be able to extend the 2D phase space to 3D through rapidly and repeatedly switching signs of constitutive parameters with remote magnetic fields. It is shown for the first time that effective modulus can be reversibly switched between positive and negative within controlled frequency regimes through lattice buckling modulated by theoretically predicted magnetic fields. The magnetically triggered negative-modulus and cavity-induced negative density are integrated to achieve flexible switching between single-negative and double-negative. This strategy opens promising avenues for remote, rapid, and reversible modulation of acoustic transportation, refraction, imaging, and focusing in subwavelength regimes.

Contributed Papers

11:10

2aSA9. Vibration suppressions using a dynamic absorber embedded with acoustic black hole features. Tong Zhou and Li Cheng (Dept. of Mech. Eng., Hong Kong Polytechnic Univ., Hong Kong 852, Hong Kong, mmlcheng@polyu.edu.hk)

By capitalizing on the acoustic black hole (ABH) phenomena, an ABH-featured vibration absorber is proposed for the broadband vibration suppressions of a primary structure. As an add-on device to be attached to the primary structure, the proposed absorber embraces the principles of both dynamic vibration absorbers and waveguide absorbers. Its design and implementation avoids the tedious parameter tuning, thus showing robustness to accommodate structural variations in the primary structure. Using a beam as a benchmark example, both numerical simulations and experiments show that multiple resonances of the primary structure can be significantly reduced, and the same absorber can be used for different primary systems. Analyses reveal the existence of three types of vibration reduction mechanisms, manifested differently and dominated by different physical process, *i.e.,*structural interaction, damping enhancement, and their combination. Comparisons with a conventional uniform beam absorber shows that the superiority of the proposed absorber is attributed to its ABH-specific features exemplified by the enriched system dynamics and the enhanced broadband damping.

11:25

2aSA10. Acoustic carpet cloaks based on anomalous reflectors. Marc Martí-Sabaté (Dept. of Phys., Universitat Jaume I, Castellón de la Plana, Spain), Yabin Jin (I2M, Univ. of Bordeaux, Bordeaux, France), and Daniel Torrent (Dept. of Phys., Universitat Jaume I, Av. de Vicent Sos Baynat, s/n, Castellon de la Plana, Castellón 12071, Spain, dtorrent@uji.es)

We present an acoustic carpet cloak based on acoustic anomalous reflectors. The cloak consists on a engineered diffraction grating, designed in such a way that a wave incident with a given angle is "retroreflected," so that a pyramidal surface reflects a normally incident wave as if it were a flat surface. Different geometries and angles of the surface are tested, showing that the surface can have an arbitrary shape, as long as the retrorreflection conditions be satisfied locally. Finally, the experimental characterization of these cloaks is discussed.

TUESDAY MORNING, 6 NOVEMBER 2018 SALON A (VCC), 8:15 A.M. TO 11:50 A.M.

Session 2aSC

Speech Communication, Biomedical Acoustics, and Signal Processing in Acoustics: Recent Advances in Experimental, Computational, and Clinical Research in Voice Production and Perception

Zhaoyan Zhang, Cochair
UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Center, Los Angeles, CA 90095

Michael Doellinger, Cochair
University Hospital Erlangen, Erlangen, Germany

Chair's Introduction—8:15

Invited Papers

8:20

2aSC1. Reconsidering the nature of voice. Jody E. Kreiman (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Voices play essential roles in human experience, and carry many kinds of meaning. At present, the study of voice is consistently tied to the speech chain model, so that voice production, acoustics, and perception are treated as separate, independently studied stages. Such studies provide no insight into how meaning in voice is constructed, and create a situation where voice comprises something different to everyone who studies it. This paper presents empirical and theoretical arguments leading to the conclusion that voice must be viewed as a single integral process, and not as a concatenation of separate stages. We argue that voice production, acoustics, aeroacoustics, biomechanics, perception, and all the rest, are conceptually inseparable, and that none can be understood without knowing how its relationship to the others contributes to its structure and function. We submit that development of such a comprehensive theory of voice should be the primary goal of voice research, whatever the discipline. Such a theory would form a basis on which to build realistic explanations of how voice functions socially and communicatively, how it has evolved, and how it conveys meanings of all sorts, leading to a deeper understanding of what it means to be human and to communicate.

8:40

2aSC2. Computational medicine in voice research. Nicole Y. Li-Jessen (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8/F, Montreal, QC H3A1G1, Canada, nicole.li@mcgill.ca)

Computer models have been widely used to numerically simulate the physical dynamics of vocal fold oscillation in phonation. Computer simulations have allowed the exploration of a much wider parameter space and a much longer time scale than what would be very costly or sometimes impossible with animal and human models. In the past 10 years, our team uses a combined *in vivo, in vitro,* and *in silico* approach to understand the biological mechanism underlying the formation of vocal lesions. The ultimate goal is to identify key biological and physical factors associated with the pathogenesis of and recovery from vocal fold injury that will be useful for personalized voice restoration. Tissue injury and repair is a complex process that involves multiple cells types and molecular signals. We developed 3D agent-based models at a physiological scale to simulate and visualize the cellular and molecular response to vocal injury and treatment. Significant progress has made to improve the model's computational speed and visualization features using our hybrid CPU-GPU computing platform. Gaps however still exist between the model and the reality. In this talk, I will present what we have achieved so far as well as what challenges we are facing in biological computing.

9:00

2aSC3. Studying vocal fold non-stationary behavior during connected speech using high-speed videoendoscopy. Maryam Naghibolhosseini, Dimitar Deliyski (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, naghib@msu.edu), Stephanie Zacharias (Laryngotracheal Regeneration Lab, Mayo Clinic, Phoenix, AZ), Alessandro de Alarcon (Div. of Pediatric Otolaryngol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and Robert Orlikoff (College of Allied Health Sci., East Carolina Univ., Greenville, NC)

Studying voice production during running speech can provide new knowledge about the mechanisms of voice production with and without disorder. Laryngeal high-speed videoendoscopy (HSV) systems are powerful tools for studying laryngeal function and, if coupled with flexible fiberoptic endoscopes, they can provide unique possibilities to measure vocal fold vibration with high temporal resolution during connected speech. Hence, we can measure the non-stationary behaviors of the vocal folds, such as the glottal attack and offset times in running speech. In this study, a custom-built flexible fiberoptic HSV system was used to record a "Rainbow Passage" production from a vocally normal female. Automated temporal and spatial segmentation algorithms were developed to determine the time stamps of the vibrating vocal folds and the edges of the vocal folds during phonation. The glottal attack time and offset times were then measured from the temporally and spatially segmented HSV images. The amplification ratio was computed during the phonation onset and the damping ratio was calculated at the offset of sustained portion of phonation. These measures can be used to describe the laryngeal mechanisms of voice production in connected speech.

9:20

2aSC4. High-fidelity image-based computer modeling of voice production—From muscle contraction to flow-structure-acoustics interaction. Qian Xue, Xudong zheng, Weili Jiang, Ngoc Pham, and Biao Geng (Mech. Eng., Univ. of Maine, 213 Boardman Hall, Orono, ME 04469, qian.xue@maine.edu)

This study aims to develop a high-fidelity computer model of voice production which can employ the image-based realistic three-dimensional geometries of larynges and simulate the complex mechanical process of voice production and control, from muscle contraction to flow-structure-acoustics interactions. Such a model will advance our understanding of the relationship between muscle contraction, vocal fold posturing, vocal fold vibration, and final voice outcome, which has important clinical implications for voice management, training, and treatment. The key components of the model include a sharp-interface-immersed-boundary-method based impressible flow model, a finite-element method based nonlinear structural dynamics model and a hydrodynamics/acoustics splitting method based acoustics model. A Hill-based contractile model is coupled in the finite element analysis to capture the active response of vocal fold tissues, and a fiber-reinforced model is employed for the passive response. A series of validations have been performed. The coupled Hill-based model and fiber-reinforced model demonstrated a good agreement with literature experimental data for dynamics, concurrent tissue stimulation, and stretching. The flow-structure-acoustics interaction model was validated with excised canine experiments using realistic geometric and material properties. The simulations showed a good agreement on the fundamental frequency, vocal fold maximum divergent angle, flow rate, and intraglottal velocity and pressure fields.

9:40

2aSC5. Bayesian inference of tissue properties from glottal area waveforms using a 2D finite element model. Paul J. Hadwin and Sean D. Peterson (Dept. of Mech. and Mechatronics Eng., Univ. of Waterloo, 200 University Ave. W, Waterloo, ON N2L 3G1, Canada, pjhadwin@uwaterloo.ca)

Bayesian inference has recently been demonstrated to be effective in estimating stationary and non-stationary reduced-order vocal fold model parameters, along with the associated levels of uncertainty, from simulated glottal area waveform measures (Hadwin *et al.*, 2016, 2017). In these studies, the fitting model was a three mass body-cover model with two degrees of freedom in the cover layer. While demonstrative, restricting the fitting model to two degrees of freedom assumes a priori that this is sufficient to capture salient vocal fold dynamics, thus limiting future clinical applicability. To overcome this, we employ Bayesian inference to directly estimate tissue properties of a two-dimensional (2D) finite element vocal fold model from glottal area waveforms generated by both numerical simulations and recorded videos of synthetic vocal fold oscillations. We demonstrate that the 2D finite element model is not only capable of producing meaningful estimates with reasonable uncertainties, but is also capable of distinguishing between different experimental tissue properties, which is an essential step towards the development of patient specific models.

10:00

2aSC6. Advancement of flow velocity measurements in excised canine larynx model. Liran Oren, Charles Farbos de Luzan, Alexandra Maddox, Ephraim Gutmark, and Sid M. Khosla (Univ. of Cincinati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

In the classic source-filter model of speech production, flow modulation is the source of sound. Modulation of the flow refers to the fact that the volume flow, Q, is changing as the glottis opens and closes. Furthermore, studies have shown that the maximum flow declination rate (MFDR), which occurs during the latter part of the closing phase, is highly correlated with acoustic intensity (loudness) and acoustic energy in the higher harmonics. Therefore, measurements of the glottal airflow can provide important insights into voice mechanisms, dysfunction and efficiency. In recent years, particle image velocimetry (PIV) have become the method of choice for measuring Q because the technique can quantify, non-intrusively, the spatial and temporal information of the flow. The discussion includes progress of PIV measurements, from 2D to tomographic measurements, in the excised canine larynx model. Key findings related to intraglottal flow and glottal geometry, and their extension to modeling of voice mechanisms, are described.

10:20–10:35 Break

Contributed Papers

10:35

2aSC7. Systematic analysis of fluid-structure-acoustic interactions in an ex-vivo porcine phonation model. Marion Semmler (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Med. School, Dept. of Otorhinolaryngology, Head & Neck Surgery, Waldstr. 1, Erlangen, Bavaria 91052, Germany, marion.semmler@uk-erlangen.de), David Berry (Dept. of Head and Neck Surgery, David Geffen School of Medicine at UCLA, Los Angeles, CA), Anne Schützenberger, and Michael Döllinger (Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Med. School, Dept. of Otorhinolaryngology, Head & Neck Surgery, Erlangen, Bavaraia, Germany)

Background: The phonation process is a complex interaction of different components, namely, the airflow from the lungs, the mucosal tissue in the larynx, and the resulting acoustic signal. From clinical experience, it is assumed that the presence of a pre-phonatory gap, oscillation asymmetries, and aperiodicities constitute detrimental factors to the resulting voice quality. In order to gain an in-depth understanding of the cause-effect chain and enable effective therapeutic measures, the crucial elements of the process need to be studied in isolation. *Materials & Methods:* The presented experimental set-up allows a systematic control and monitoring of the individual components of the excised porcine larynx. Varying degrees of asymmetric adduction, pre-phonatory gap size and flow rate reproduce irregular oscillation patterns associated with pathologic voice production. *Results:* The statistical analysis based on k-means clustering allows the identification of structural and aero-dynamic factors which influence voice quality. In addition to an increase in glottal closure and oscillation symmetry, an increase in glottal flow resistance produced a beneficial effect on the acoustic output signal and the closely coupled subglottal pressure. *Conclusion:* The clinical applicability and therapeutic value of these insights must be assessed in further studies.

10:50

2aSC8. The importance of medial surface vertical thickness to the control of voice production. Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

While vocal fold vibration is often studied from a superior view, the vertical thickness of the vocal fold medial surface (or the portion that forms the glottal constriction) has long been hypothesized to play an important role in the control of voice production. This importance of the vertical thickness is confirmed in our recent computational studies, which showed that the vertical thickness, not the degree of vocal fold approximation, has a dominant effect on the vocal fold contact pattern and the spectral shape of the produced voice. Specifically, thicker folds often vibrate with a larger closed quotient, a lower H1-H2 value, and stronger excitation of higher-order harmonics. Thicker folds are also more likely to exhibit irregular vibration. These studies also showed an important effect of the transverse stiffness of the vocal fold in the coronal plane on the resulting voice quality, but a relatively small effect of the longitudinal tension on voice quality except around phonation onset. These results suggest that in addition to the superior view, more attention should be given to vocal fold shape along the vertical dimension and its control. [Work supported by NIH grant R01DC011299.]

11:05

2aSC9. Relationship between intraglottal geometry, vocal tract constriction, and glottal flow during phonation of a canine larynx. Charles Farbos de Luzan, Sid M. Khosla, Liran Oren, Alexandra Maddox (Dept. of Otolaryngol. - HNS, Univ. of Cincinnati, 231 Albert Sabin way, MSB 6313, Cincinnati, OH 45267-0528, farboscs@ucmail.uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

Hypothesis: The rapid reduction of the glottal volume flow rate, which usually occurs in normal phonation during the closing of the vocal folds, is measured by a quantity known as the maximum flow declination rate (MFDR). Our preliminary work highly suggests that intraglottal flow separation vortices (FSV), which form near the superior aspect of the divergent folds during closing, can directly affect MFDR. In this project, we hypothesize that the strength of the vortices is highly correlated with higher MFDR, larger divergence angles and stronger FSV. *Methodology*: We use particle image velocimetry to measure the strength of the FSV as well as the intraglottal pressures during phonation in four excised canine larynges with an attached mechanical vocal tract. Two vocal tract models are used, one with a false fold gap of 7 mm, one with a gap of 3 mm. *Results*: Stronger FSV (measured by vorticity) are correlated with glottic efficiency, larger glottal divergence angles and/or with decreased distance between the false folds. *Conclusion*: We show correlation of the strength of the FSV with different variables. These data are currently being used to validate computational flow structure models. These computational models then can be used to determine causation.

11:20

2aSC10. Multitaper harmonic analysis of infant vocalizations. Gordon Ramsay (Dept. of Pediatrics, Emory Univ. School of Medicine, Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, gordon.ramsay@emory.edu)

Infant vocalizations exhibit many special complexities relative to adult speech that make acoustic analysis difficult, even though the basic physics of sound production in the vocal tract is identical. Fundamental frequency is raised, resulting in wider-spaced harmonics that make it hard to accurately estimate the spectral envelope or locate formant resonances, which are higher than in adult voices, due to anatomical scaling effects. Irregular vocal fold vibratory regimes are often observed, involving period doubling and other nonlinear phenomena not seen in adult speech, along with atypical patterns of turbulent noise production. Unsurprisingly, traditional analysis techniques that depend on estimating the spectral envelope from the speech signal do not work well when applied to infant vocalizations. An alternative approach is proposed, using multitaper analysis to calculate the time-varying amplitude and phase of every harmonic component of the voice, along with the residual noise component, recovering the vocal tract transfer function from the results. The new technique is compared against linear

prediction and cepstral analysis using home audio recordings of 20 infants collected from 0 to 24 months using LENA technology. Developmental progressions in the acoustic structure of the infant voice are identified that cannot be found using traditional methods.

11:35

2aSC11. Adult imitating child speech: A case study using 3D ultrasound. Colette Feehan (Linguist, Indiana Univ., Bloomington, 107 s Indiana Ave., Bloomington, IN 47404, cmfeehan@indiana.edu)

Voice actors are interesting for linguistic study because of their unique abilities to manipulate their vocal tract and convey different social identities. This is essentially a field of professional folk linguistics where professionals manipulate their vocal tracts to convey socially indexed, linguistic features. This study uses Ultrasound paired with acoustic analyses to address what one amateur voice actor does while imitating a child voice. Previous studies have looked at anatomical and acoustic variations defining different character types such as laryngeal setting (Teshigawara, 2003; Teshigawara & Murano, 2004) and breathy voice in Anime (Starr, 2015). This study addresses specific tongue morphology resulting from an adult imitating a child's voice and serves as a pilot for future study of professional actors. The participant is one adult, amateur actor who produced CV syllables at different places of articulation with different vowel qualities. Notable manipulations are hyoid bone raising, gesture fronting, and tongue "troughing" where the sides of the tongue are used to narrow the oral cavity. The actor constricts the filter in multiple ways to shrink the usable space in the oral cavity and imitate the acoustic signal from a child's vocal tract. This presentation explores the anatomical manipulations the actor utilizes.

TUESDAY MORNING, 6 NOVEMBER 2018 OAK BAY 1/2 (VCC), 8:45 A.M. TO 11:45 A.M.

Session 2aUW

Underwater Acoustics: Signals and Systems I

Danelle E. Cline, Chair
R&D, MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039

Contributed Papers

8:45

2aUW1. Deconvolved conventional beamforming applied to the SWellEx96 data. Tsih C. Yang (Ocean College, Zhejiang Univ., Bldg. of Information Sci. and Electron. Eng., 38 Zhe Da Rd., Hangzhou, Zhejiang 310058, China, tsihyang@gmail.com)

Horizontal arrays are often used to detect/separate a weak signal and estimate its direction of arrival among many loud interfering sources and ambient noise. Conventional beamforming (CBF) is robust but suffers from fat beams and high level sidelobes. High resolution beamforming such as minimum-variance distortionless-response (MVDR) yields narrow beam widths and low sidelobe levels but is sensitive to signal mismatch and requires many snapshots of data to estimate the signal covariance matrix, which can be a problem for a moving source. Deconvolution algorithm used in image de-blurring was applied to the conventional beam power of a uniform line array (spaced at half-wavelength) to avoid the instability problems of common deconvolution methods and demonstrated with real data (T. C. Yang, IEEE J. Oceanic Eng., 43, 160–172, 2018). The deconvolved beam output yields narrow beams, and low sidelobe levels similar to MVDR and at the same time retains the robustness of CBF. It yields a higher output signal-to-noise ratio than MVDR for isotropic noise. The method is applied here to the horizontal array data collected during the SWellEx96 experiment. Bearing time record are created to compare the performance of various beamforming methods and used to track the source position.

9:00

2aUW2. Using convolutional neural networks on perceptual and spectral features: Classification experiments on the TREX'13 dataset. Tara J. LeBlanc and John Fawcett (Defence Res. and Development Canada, Atlantic, 9 Grove St., P.O. Box 1012, Dartmouth, NS B2Y 2E2, Canada, tara.leblanc@drdc-rddc.gc.ca)

Distinguishing objects of interest on the seafloor from clutter remains a key problem facing the automatic target recognition (ATR) community. Because scattering at high frequencies relates more to the geometry of the scatterer, traditional ATR on high frequency imaging sonars is known to suffer in areas of high clutter. Recently, there has been interest in low frequency (1–50 kHz) wideband imaging sonars, because the complex combination of external (geometric) and internal (elastic) scattering that occurs in this frequency range is thought to improve classification in areas of high clutter. This work investigates the classification problem of distinguishing unexploded ordnance (UXO) from clutter using image-based Convolutional Neural Networks on the TREX'13 dataset. This dataset consists of experimental and modelled acoustic backscatter for objects interrogated acoustically at 3–30 kHz across a range of aspects. The model data are used exclusively for training, and the experimental data are used exclusively for testing purposes. The generated feature-set is a combination of acoustic colour and perceptual features, which are derived from modelling how humans perceive timbre. Further in an effort to address the differing amplitude modulations between sets of model and experimental data and improve classified performance, several moving window normalizations will be investigated.

9:15

2aUW3. Out-of-band beamforming in shallow water with horizontal arrays. Alexander S. Douglass (Mech. Eng., Univ. of Michigan, 2010 AL, 1231 Beal, Ann Arbor, MI 48109, asdougl@umich.edu), Shima Abadi (Mech. Eng., Univ. of Washington, Bothell, Bothell, WA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Out-of-band beamforming methods, frequency-difference and frequency-sum beamforming, are array signal processing methods that allow a user to process signals at either a lower, below-the-signal-band difference frequency, or higher, above-the-signal-band sum frequency. Frequency-difference beamforming has previously been investigated for shallow-ocean propagation with sparse vertical arrays, demonstrating its capability to overcome the negative effects of spatial aliasing [Douglass, Song, and Dowling (2017). J. Acoust. Soc. Am. **142**, 1663–1673]. In the work presented here, data from the 2012 COAST (Cascadia Open-Access Seismic Transects) experiment is utilized to explore out-of-band methods with horizontal arrays, where variations in environmental characteristics (bathymetry, sound speeds, etc.) may be more pronounced across the array. The abundance of array elements (636 in total) and total array length (7.9 km) in this experiment provide a means for controlling the sparsity and signal coherence within a chosen subarray to determine the capabilities and limitations of out-of-band beamforming. In addition, the bandwidth of the signal of interest is sufficient to compare out-of-band methods to conventional methods where the difference or sum frequencies match the in-band frequency used with conventional beamforming. [Sponsored by ONR.]

9:30

2aUW4. Underwater acoustics array spatial reversal convolution and beamforming. Anbang Zhao, Lin Ma, Chunsha Ge, Xuejie Bi (Acoust. Sci. and Technol. Lab.,Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University),College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, HLJ 150001, China), (malin@hrbeu.edu.cn)

A novel underwater acoustic array signal processing method based on spatial reversal and convolution, aiming at improving detection performance, is proposed to meet the engineering application demand of the underwater acoustic array detection system. The theoretical and mathematical model of the proposed method is deduced and presented in detail. Based on the extended covariance matrix of the physical array, the array sensitivity equalization and the noise suppression technique of the underwater acoustic array are studied. The additional gain provided by sensitivity equalization and noise suppression techniques is analyzed and exploited. Computer modelling and various numerical simulations are applied to demonstrate the proposed method's effectiveness. At-sea experimental data are processed to test the performance in real underwater acoustic environment. These results indicate that the proposed array processing method can effectively improve the detection ability. The proposed techniques in this paper provide an insightful way to improve the detection performance of the underwater acoustic array.

9:45

2aUW5. Adaptive beamforming for uniformly-spaced linear hydrophone array using temporal convolutional neural networks. Dezhi Wang, Lilun Zhang, Changchun Bao (College of Meteorol. and Oceanogr., National Univ. of Defense Technol., Sanyi St., Kaifu District, Changsha 410073, China, wang_dezhi@hotmail.com), Kele Xu (School of Comput., National Univ. of Denfense Technol., Paris, France), Boqing Zhu (School of Comput., National Univ. of Denfense Technol., Changsha, China), and Zengquan Lu (College of Meteorol. and Oceanogr., National Univ. of Defense Technol., Changsha, China)

In oceanic remote sensing, large discrete linear hydrophone arrays are usually utilized to enhance the signal-to-noise ratio (SNR) by means of beamforming that attenuates noise coming from the directions outside the target direction. Although the widely adopted delay-and-sum (DS) and filter-and-sum (FS) beamforming techniques can improve the performance of linear hydrophone arrays in some scenarios, both DS and FS methods have limited adaptability to the changing oceanic environments especially with spatially correlated noise. Moreover, these beamforming techniques aims to optimize the SNR, which is not completely consistent with some objectives such as target recognition. In this work, a neural network adaptive beamforming framework for an uniformly-spaced linear hydrophone array is proposed to make use of the large-scale array signals and address the above issues. In particular, an architecture consists of temporal convolutional neural networks is designed to predict the beamforming filter coefficients in time domain by taking the raw multi-channel waveforms of sound pressure as the input. The filter prediction networks are also jointly trained with a convolutional neural network (CNN) based classification model with the purpose of increasing the target recognition accuracy. The proposed approach is validated by experiments carried out in the south coastal waters of China.

10:00–10:15 Break

10:15

2aUW6. Detection and classification of whales calls using band-limited energy detection and transfer learning. Danelle E. Cline and John P. Ryan (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, dcline@mbari.org)

The Monterey Bay Aquarium Research Institute has been recording since July 2015 almost continuously at the Monterey Accelerated Research System (MARS) cabled observatory in Monterey Bay, California, USA. This long-term recording contains thousands of whale calls to help further our understanding of interannual, seasonal, and diel patterns. Here we report on our highly accurate detection and classification method developed to classify blue whale A, B, and D calls and Fin whale 20 Hz pulses. The foundation of the method is a computationally efficient and tuned decimation filter to convert the broadband hydrophone 128 kHz signal to 2 kHz which preserves the low-frequency signal and avoids any high-frequency aliasing. Detection is done using a band-limited-energy-detection filter to find potential calls in the decimated data. Spectrograms are then generated for potential calls and enhanced with local image normalization followed by smoothing by convolving in either time or frequency. Classification is done using the Google Inception v3 model with a transfer learning method. Overall, false positive rates are very low despite variability in whale call shape and background noise.

10:30

2aUW7. Passive source depth discrimination in deep-water. Rémi Emmetiere (ENSTA Bretagne, 2 rue Francois Verny, Brest Cedex 9 29806, France, remi.emmetiere@ensta-bretagne.org), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Marie Gehant Pernot (Thales Underwater Systems, Sophia Antipolis, France), and Thierry Chonavel (IMT Atlantique, Plouzané, France)

In this work, we address the problem of passive source depth discrimination using a horizontal line array (HLA). The scope is restricted to low-frequency sources ($f<500$ Hz), broadband signals (a few Hertz bandwidth), deep-water environment ($D>1000$ m), and distant sources (at least several kilometers). The proposed method is based on the underlying physics driving the propagation. It notably uses the concept of mode trapping and waveguide invariant. In deep water, the waveguide invariant largely depends on source depth, and is thus an interesting input for source depth discrimination. An algorithm is proposed to compute energy ratio in modal groups. The input data for the algorithm are a range-frequency intensity, as measured on a HLA. The modal groups are thus defined based on their respective waveguide invariant values, which is a main difference with existing depth discrimination methods adapted to shallow water context. This idea is explored, and extended to propose a source depth discrimination which is performed as a binary classification problem. As long as the sound speed profile features a surface thermocline, the algorithm does not require detailed knowledge about the environment and it allows one to classify sources under two hypotheses, above or under a threshold depth.

10:45

2aUW8. A Bayesian approach to a passive range estimation method of underwater sound source in shallow water waveguide. Xiaoman Li and Shenchun Piao (Harbin Eng. Univ., No.145, Nantong St., Nangang District, Harbin, Heilongjiang, Harbin 150001, China, lixiaoman@hrbeu.edu.cn)

In this paper, a matched mode processing method is used to estimate the range of underwater sound source in shallow water, in which the replica field is calculated with the P and Q normal mode model. In this normal mode model, the replica field is described with the bottom phase shift coefficient, the sea depth and the average sound speed profile. The influence caused by the uncertainty of environment parameters, on the accuracy of range estimation is discussed with Bayesian approximation. Bayesian approximation is a good way to estimate the unknown parameters based on probability of data. The Gaussian distribution model and Markov Chain Monte Carlo method are used in this paper for calculating the posterior probability density. The effect of the uncertainty or error of environment parameters to the ranging result is obtained. The practicality of the ranging method is raised significantly. The procedure and the discussed results are given.

11:00

2aUW9. Experimental performance evaluation of underwater active detection and positioning system. Lin Ma, Anbang Zhao (Acoust. Sci. and Technol. Lab.,Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University),College of Underwater Acoust. Eng., No. 145, Nantong St., Harbin, HLJ 150001, China, malin@hrbeu.edu.cn), T. Aaron Gulliver (Dept. of Elec. & Comput. Eng., Univ. of Victoria, Victoria, BC, Canada), and Caigao Zeng (Acoust. Sci. and Technol. Lab., Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University), College of Underwater Acoust. Eng., Harbin, China)

Underwater targets active detection and positioning are of increasing challenges in various underwater practical engineering applications. The performance of the proposed detection and positioning algorithms are validated and evaluated by designed open-lake experiments. The widely used matched-filter, conducted in frequency-domain, is improved by normalized least mean square algorithm to achieve the better detection ability. The improved detector is applied to the underwater targets detection in these experiments. Post-processing technique based on autocorrelation is proposed to enhance the performance of detection by cancelling the noise existing in frequency-domain. The underwater target is located on the basis of the time-delay and azimuth estimation. The open-lake experimental results show that the detection and positioning system is robust and effective in the complex underwater environment. The designed model and these proposed techniques provide a better way to underwater target active detection and positioning.

11:15

2aUW10. Optimization of deployment depth for active towed array sonar (ATAS) using simulated annealing. Sangkyum An (Ocean Eng., Seoul Univ., Dept. of Naval Architecture and Ocean Eng., Collage of Eng., Seoul National Univ., Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, navy60@snu.ac.kr), Keunhwa Lee (Defense Systems Eng., Sejong Univ., Seoul, South Korea), and Woojae Seong (Ocean Eng., Seoul Univ., Seoul, South Korea)

The active towed array sonar (ATAS) is an active operating sonar towed behind the surface ship and deployed at various depths. The performance of ATAS may vary with the deployment depth. This study proposes a method which calculates the optimal deployment depth using simulated annealing (SA) in order to reduce the total computational time spent in the direct search of the deployment depth. We define a sonar performance function (SPF) with the probability of detection, which represents as a degree of how well the ATAS might perform at particular deployment depth. Then, the optimal depth is defined as the depth where the SPF is maximized. The SPF depends on the acoustic environment, target position and source-receiver depth. The optimization results are compared with the direct calculation of SPF at all deployment depth of source and receiver. Also, our study provides the optimal depths in the area of East Sea of Korea.

11:30

2aUW11. Bi-directional equalization for long-range underwater acoustic communication in East Sea of Korea. Hyeonsu Kim, Sunhyo Kim, Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 15588, South Korea, hskim00@hanyang.ac.kr), and Ho Seuk Bae (The 6th R&D Inst., Agency for Defense Development, Changwon, South Korea)

The long-range underwater acoustic communications using a sound channel in deep sea have been studied for many years. The acoustic waves transmitted through the sound channel can be propagated by long distance because there is no interaction with the sea surface and bottom interfaces. However, the transmitted signals are distorted by multipath propagation. Time reversal combining followed by a single channel equalizer can reduce efficiently the effects of multipath communication channel and thus increases signal-to-noise ratio by obtaining temporal and spatial diversity. However, the conventional time reversal tends to increase the entire error rate when burst errors occur. Long-range communication signals are particularly vulnerable to burst errors due to the weak signal strength. In this talk, the bi-directional equalization is applied to compensate for the distortion of long-range communication signals where the burst errors occur. Since the bi-directional equalization combines the soft outputs of forward and backward equalization, the effect of error propagation due to burst error can be reduced. The long-range communication experiment data acquired in the East Sea of Korea in October 1999 is used to verify the performance of the bi-directional equalization. The results show that the bi-directional equalization can be effectively applied when burst errors exist. [Work supported by the ADD(UD170022DD) and the National Research Foundation of Korea (NRF-2016R1D1A1B03930983).]

TUESDAY AFTERNOON, 6 NOVEMBER 2018 THEATER (VCC), 1:00 P.M. TO 4:15 P.M.

Session 2pAA

Architectural Acoustics and ASA Committee on Standards: Auditorium Acoustics and Architectural Design: Challenges and Solutions II

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

Thomas Scelo, Cochair
Marshall Day Acoustics, 1601, 16/F The Hollywood Centre, 233 Hollywood Road, Sheung Wan 0000, Hong Kong

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. One project begets many: The role of one seminal project in the development of many design solutions (Part 1). Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

This paper, presented in three parts, describes the role of Threshold Acoustics' study and corrective treatments to Verizon Hall at the Kimmel Center for the Performing Arts, as a seminal project in the firm's work. Elements of the study of and solutions developed to improve the hall have proven to be repetitive themes in our work, both in the renovation of other halls and in the design of new spaces. This initial paper describes the interactions with the Philadelphia Orchestra, and the how the subjective concerns expressed by the orchestra led to unexpected conclusions regarding the overhead canopy and walls at the downstage edge. This portion of the presentation will focus on lessons learned for the interactions between the stage canopy weight, spatial density, and height and how these have informed subsequent design work.

1:25

2pAA2. One project begets many: The role of one seminal project in the development of many design solutions (Part 2). Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

In the second part of this multi-part paper, the solution for the cap of the organ chamber at Verizon Hall will be presented. The original cap was a lightweight material, allowing significant low frequency energy—critical to the success of any pipe organ—to escape the acoustic volume of the room. The solutions implemented to improve the sound of the organ were limited by structural capacities, and were ultimately developed as lightweight but very stiff honeycomb core panels. The strategy of lightweight-yet-stiff materials has been implemented in numerous other projects to achieve low frequency reflectivity. Examples will be presented illustrating development of this strategy over the past decade.

1:45

2pAA3. One project begets many: The role of one seminal project in the development of many design solutions (Part 3). Gregory A. Miller and Scott D. Pfeiffer (Threshold Acoust., LLC, 141 W. Jackson Boulevard, Ste. 2080, Chicago, IL 60604, gmiller@thresholdacoustics.com)

In the third and final part of this multi-part paper, computer modeling will be described as a tool for presentation to client groups (often skilled listeners but not expert acousticians). Over the course of the Verizon Hall project, computer models were used to present visual illustrations that explained the acoustic concerns experienced by the orchestra. Starting with this project, Threshold began to develop computer models as tools to translate acoustic phenomena in ways that clients could understand visually and, over time, aurally. The development of customized computer modeling techniques has further supported communication with non-acousticians, and examples of these techniques will be presented.

2:05

2pAA4. Vibration isolation for world-class performance spaces, Part I: A brief history. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

World-class performance spaces are typically located in central urban areas, often surrounded by sources of groundborne noise such as subways and streetcar lines or sources of structureborne noise within the building itself which, if not addressed, can be disruptive to the performances inside the buildings. The advent of digital recording within performance spaces has also been a major motivation for reducing background noise, often to levels below the threshold of human perception. The history and development of vibration isolation incorporated into such spaces will be discussed.

2:25

2pAA5. Vibration isolation for world-class performance spaces, Part II: Examples. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

This paper will present examples of vibration isolation incorporated into world-class performance spaces in North America and oversees, including acoustical and vibration measurements demonstrating the effectiveness of the vibration isolation to reduce groundborne/structureborne noise.

2:45–3:00 Break

3:00

2pAA6. Identifying trends in worship space acoustics: 3 decades of design. Stephanie A. Ahrens, Erica E. Ryherd, Lauren M. Ronsse (Architectural Eng., Univ. of Nebraska - Lincoln, Durham School of Architectural Eng. & Construction, Omaha, NE 68182-0816, sahrens@unomaha.edu), and David T. Bradley (Inclusion, Diversity and Equity, Smith College, Northampton, MA)

Worship Space Acoustics: 3 Decades of Design—published by the Acoustical Society of America in 2016—is a book rich with detailed acoustic and architectural information for 67 worship venues with diverse acoustic designs. The venues represent a range of worship space types, various geographic locations around the globe, an assortment of architectural typologies, and a variety of sizes (including seating capacities ranging from 100 to 21,000). This project aimed to synthesize the acoustic and written content presented in the book in order to identify design trends across the 67 venues. Qualitative data included venue typology, location of materials, sound system design, floor plate and room shape, intended use of space, and noise sources, which were mined and analyzed using text analytics software. Quantitative data included reverberation time, background noise level, seating capacity, and room volume. By compiling the various qualitative and quantitative data across all venues, acoustic data and design strategies were synthesized to illustrate popular trends in worship space acoustic design over the last three decades.

3:15

2pAA7. The modern sophistication of acoustics in education environments. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

In the past 10 years the use of acoustic design and products in educational environments, from elementary schools to universities, has grown exponentially. While acoustics in general has been more widely embraced to solve problems related to modern design technologies, the use of more sophisticated designs and materials has been widely accepted as an alternative to the simple absorptive materials of the past. Some secondary school designs challenge the finest college and university campuses for both creative and effective acoustic spaces. Likewise, some colleges and universities challenge the finest commercial venues, studios and rehearsal spaces in both function and design. Examples will be shown in the presentation including a new convertible acoustic system at Tabor College designed to make the hall usable for both classical performance and sound reinforced, high volume contemporary performances as well. Discussions as to the principals behind the use of acoustics in these environments will be offered.

3:30

2pAA8. The preference of orchestra ensemble on sound absorption design in a concert stage. Yi Run Chen (Architecture, National Taiwan Univ. of Sci. and Technol., No.43, Keelung Rd., Sec.4, Taipei 10607, Taiwan, D10413009@mail.ntust.edu.tw), Lawrence Huang (none, Kaohsiung, Taiwan), and Weihwa Chiang (Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

Musicians in a symphony orchestra have to deal with more sound balance problems in a relatively compact stage when appropriate spacing between orchestra sections become unavailable. Under such circumstances, applying sound absorptive surfaces around the stage might benefit the condition of either hearing oneself or hearing each other. This paper explored the musicians' preference on energy-restrained design with varying frequency characteristics, amount of coverage and location. Experiments took place with a well-reputed university orchestra at their rectangular concert hall in the routine practices. Room acoustic parameters were measured on site and in an 1/10 scale model, empty and occupied, to be compared with computer simulations.

3:45

2pAA9. Case study: Retrofitted solution to address sound flanking via window mullions of stacked residences. Pier-Gui Lalonde (Integral DX Eng. Ltd., 907 Admiral Ave., Ottawa, ON K1Z6L6, Canada, pier-gui@integraldxengineering.ca)

This paper presents a case study concerning a noise transmission issue between stacked residences in a condominium building. The façade design included vertical window mullions that were continuous and common to the upper and lower units, creating a sound flanking path. Low acoustic privacy and disturbing intrusive noise resulted. A solution was required to mitigate the noise transmission issue while meeting additional requirements: fully reversible modifications to façade components were preferred, no interference to the function of the façade components was allowed, and occupant expectations regarding the appearance of the final solution needed to be met. This paper will present the approach that was followed in order to diagnose the issue, as well as the analysis which led to the solution that was ultimately implemented.

4:00

2pAA10. Noise and vibration measurements of a machine-room-less elevator system. Andrew Williamson (RWDI, 301-2250 Oak Bay Ave., Victoria, BC V8R 1G5, Canada, andrew.williamson@rwdi.com)

A trend in the design of new condominium buildings is the use of "machine-room less" (MRL) elevator systems. The primary difference between MRL elevator systems and more traditional elevator systems is that, rather than locating the hoist machinery in a rooftop penthouse, the hoist machinery is installed in the elevator shaft. In this location (which is typically closer to residential units), there is greater potential for the operation of the machinery to result in complaints from residents of excessive noise and vibration levels. This paper presents a case study of noise and vibration measurements that were conducted in a condominium building with a MRL elevator system. Measurements were conducted both within the affected residential unit as well as in the elevator shaft. In conducting the measurements, particular attention was paid to determining the effectiveness of the vibration isolation that was provided for the elevator hoist machinery within the elevator shaft.

Session 2pAB

Animal Bioacoustics and Signal Processing in Acoustics: Anything You Can Do I Can Do Better: Bat Versus Dolphin Biosonar

Laura Kloepper, Cochair
Biology, Saint Mary's College, 262 Science Hall, Saint Mary's College, Notre Dame, IN 46556

Brian K. Branstetter, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106

Chair's Introduction—1:00

Invited Papers

1:05

2pAB1. The acoustic world of bat biosonar. Rolf Müller, Michael J. Roan, Mohammad Omar Khyam, and David Alexandre (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Bats and toothed whales (odontocetes) have both independently evolved sophisticated biosonar systems. This raises the question how similar the functional principles of these systems are. Could, for example, insights gained from bats be assumed to hold for odontocetes or vice versa? Could both systems be lumped together as a single source of inspiration for novel engineering approaches to sonar sensing? Similarities and differences between the biosonar systems of bats and odontocetes are likely to depend on the respective acoustic environments in which these systems have evolved as well as on the evolutionary starting points and capabilities for adaptations of these two very different phylogenetic groups. In this presentation, the focus will be on comparing the acoustic environments of bats and odontocetes. The acoustics of biosonar sensing can be organized into three different aspects: (i) the properties of the propagation medium, (ii) the geometry and material of the boundaries that limit the propagation channel (this includes targets of interest and clutter), and (iii) the time-frequency and spatial properties of the sources. In this presentation, these aspects will be reviewed for the in-air biosonar of bats. In the companion talk, the same will be done for the underwater biosonar of the odontocetes.

1:25

2pAB2. Jittered echo delay resolution in bats and dolphins. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Ryan A. Jones, Jason Mulsow, Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and Patrick Moore (National Marine Mammal Foundation, Spring Valley, CA)

Fine-scale echo delay resolution has been investigated using a "jittered" echo paradigm, where animals discriminate between electronic echoes with fixed delay (i.e., simulating fixed range) and echoes with delays that alternate (jitter) on successive echoes. The data consist of the animals' discrimination performance (e.g., error rate) as a function of the amount of jitter in the echo delay (the time interval over which the echoes jitter). Results of jitter delay experiments in big brown bats (*Eptesicus fuscus*) are extraordinary (and controversial) because they suggest that bats can extract information from within the envelope of the crosscorrelation function between an emitted signal and its received echo and therefore may be capable of operating as a coherent receiver. Recently, the jitter delay paradigm has been adapted for use with bottlenose dolphins (*Tursiops truncatus*) at jitter delays down to 1 μs. Jitter delay acuity results for dolphins show qualitative similarities to those from bats: error function peak widths are below the envelope of the biosonar pulse autocorrelation function and dolphins can perceive a jitter in echo polarity with no change in echo delay. However, further testing with submicrosecond jitter values are required to determine if dolphins possess "hyperacuity" as reported for bats.

1:45

2pAB3. Reconstruction of acoustic scenes in biosonar. Chen Ming (Neurosci., Brown Univ., 814 Cascade Ct., Blacksburg, VA 24060, chen_ming@brown.edu), Michael J. Roan, Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Biosonar mechanisms are highlighted by comparing acoustically reconstructed scenes derived from generalized methods with the performance of echolocating animals. Bioinspired computations replacing standard methods then offer a path towards understanding the animal's solutions. Seemingly simple outdoor spaces present flying bats with complex scenes when vegetation and the ground are factored in, and even sound-treated research spaces such as flight rooms evolve into complex scenes as transmitted sounds propagate through the space. Underwater scenes, particularly for shallow water, are much more complicated because longer propagation distances create multipath reverberation that often overlaps with echoes. Interpulse intervals of biosonar emissions must be short to support rapid updating of tracking and perception, but reverberation means that pulse-echo ambiguity occurs. Two reconstructive methods are

useful—the HARPEX method for visualizing the progression of echoes and reverberation following a sound, and a HARPEX-like method reconfigured as a forward-looking sonar. Acoustic datasets visualize airborne and underwater sonar scenes by reconstructing sound images for successive time frames following a transmitted sound using a tetrahedral soundfield microphone or a forward-looking sonar head. The complex acoustics of vegetation are examined with a simulation model built to understand the bats' perception of vegetation. (Work supported by ONR.)

2:05

2pAB4. Biosonar-mediated prey tracking in air and in water—The fast and the persistent. Danuta M. Wisniewska (Ctr. d'Etudes Biologiques de Chizé, Ctr. National de la Recherche Scientifique, 405 Rte. de La Canauderie, Villiers-en-Bois 79360, France, danuta@stanford.edu), Mark Johnson (ZooPhysiol., BioSci., Aarhus Univ., St. Andrews, United Kingdom), Heather Vance (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St Andrews, St. Andrews, United Kingdom), Laura Stidsholt, and Peter T. Madsen (ZooPhysiol., BioSci., Aarhus Univ., Aarhus, Denmark)

Microchiropteran bats and toothed whales are the only taxa to echolocate for prey. The two biosonar systems have evolved independently, in media with very different physical properties that determine the production, propagation, and reflection of sound, as well as the means and speeds with which the echolocating predators and their prey can move and maneuver. Yet, they have converged on remarkably similar acoustic behavior; the animals adjust the timing and level of their sound emissions as they near their targets, eventually entering the buzz phase with high rates of low-amplitude signals. But while the buzz of a bat lasts a fraction of a second, prey pursuits by toothed whales are one to two orders of magnitude longer. Here, we explore these differences and their consequences for biosonar performance and prey capture success. We use echograms recorded with acoustic dataloggers on porpoises to show that, like some bat species, porpoises hunt in different environments targeting several types of prey and adjusting their biosonar outputs to manage different acoustic soundscapes. However, we further show that, unlike bats, porpoises are able to carefully track their prey with adjusted click intervals and dynamically accommodate prey movements to provide high spatial resolution without range ambiguities.

2:25–2:40 Break

Contributed Papers

2:40

2pAB5. Flexibility means adaptability: Bats adapt to jamming scenarios better than dolphins. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Bats and dolphins both use sonar to image their environment, but frequency modulated (FM) bats demonstrate greater flexibility in echolocation than their marine counterparts, modifying the time-frequency shape of their calls. Bats' FM signals can change depending on foraging phase, environment, or even density and proximity of conspecifics. By changing their time-frequency call structure, bats can acquire different information than by using a stereotyped signal, such as the impulsive signal produced by dolphins. In my talk, I argue that bats have the superior sonar system due to this flexibility. I compare the signals between bats and dolphins and demonstrate new data on how both bats and dolphins adjust echolocation signals in jamming scenarios.

2:55

2pAB6. Biosonar capabilities of large-brained dolphins and small-brained bats: Size does matter. Kaitlin R. Van Alstyne, Brian K. Branstetter, Sam H. Ridgway (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, katie.vanalstyne@nmmpfoundation.org), James J. Finneran, and Mark J. Xitco (US Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA)

The large encephalized brain of the bottlenose dolphin, with a small corpus callosum, functions to maintain continuous, 24-hour biosonar signal processing. Bats are not faced with the dolphin's challenge of maintaining constant vigilance, as bats are able to retreat to roosts for safety from predators. Although both bats and dolphins echolocate, dolphins face greater challenges in the aquatic environment. Dolphins must locate prey of a similar density to that of the environment, which itself has a high speed of sound transmission. Bats can readily distinguish between prey from the surrounding air because of a strong impedance mismatch, and the speed of sound is much slower. Perhaps stemming from these challenges, dolphins have developed a highly sophisticated echolocation system in which they are able to emit signals and receive echoes, decide, and respond accordingly at a remarkably rapid speed. Dolphin brains may also be specially equipped for early auditory development in utero. The dolphin body has a close impedance match to water, exposing the fetus to environmental sounds. Furthermore, auditory fibers develop myelin early during gestation. Thus, although dolphins and bats both echolocate, the dolphin's experience during life in the ocean promoted the development of a large and specialized brain.

3:10

2pAB7. Don't bring a knife to a gunfight: Humpback whale biosonar reigns supreme. Eduardo Mercado (Dept. of Psych., Univ. at Buffalo, SUNY, Buffalo, NY 14260, emiii@buffalo.edu)

Stroboscopic echo imaging of the sort used by most bats and dolphins is well suited for the task of rapidly gulping prey. As an information seeking strategy, however, strobing is about as sophisticated as duct-taping a flashlight onto a helmet. Humpback whale biosonar signals, in contrast, create a temporally-extended echoic soundscape the likes of which no strober could ever conceive. By spectrally interleaving quasi-CF units with packets of shorter-duration FM units, humpback whales can likely detect multiple targets simultaneously from multiple bearings, including targets more than 5 km away, while at the same time tracking movements of individual targets. By maintaining relatively stationary positions and depths while signaling, humpback whales can potentially use not only direct echoes from targets, but also modulations in familiar acoustic scenes ("acoustic glints") to interpret the movements of distant targets. By modulating temporal substructure within fixed-rate streams of signals, humpback whales may be able to extract self-generated echoes from a sea of potential maskers. The scope of vocal control, auditory processing, perceptual flexibility, and echoic memory required for a whale to construct and interpret such rich percepts far exceeds the kinds of delay-tuned, chirplet detection that bats and dolphins rely on to bump into food.

3:25

2pAB8. Welcome home: Brazilian free-tailed bats adjust their echolocation calls to detect cave opening. Kathryn McGowan and Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, kmcgowan01@saintmarys.edu)

While both bats and dolphins use sonar to navigate their environment, the sensory challenges between the two are not equal since dolphins typically navigate in the open ocean and bats often navigate in cluttered environments. Brazilian free-tailed bats (*Tadarida brasiliensis*), for example,

adjust their frequency modulated (FM) call structure based upon their environment, producing lower-bandwidth calls in open environments, and higher-bandwidth calls in cluttered environments. In this study, we examined how Brazilian free-tailed bats change the bandwidth of calls when locating the cave opening in a flat, non-cluttered environment. We extracted individual echolocation calls from two locations around the cave, one away from the cave opening and one next to edge of the cave opening, with the only distinctive difference in the environment of the locations being the presence of the cave edge. We found higher starting and lower stopping frequencies for the FM calls at the cave edge, resulting in an increased bandwidth. This higher bandwidth suggests the bats may rely on edge detection to locate the cave opening and change the bandwidth of their signals to improve target resolution when returning to the roost.

3:40

2pAB9. Dolphins' echolocation based on reverberations through skull?
Lapo Boschi (ISTEP, Sorbonne Univ., 4 Pl. Jussieu, Paris 75005, France, lapo.boschi@upmc.fr), Michael Reinwald, Quentin Grimal (Biomedical Imaging Lab., Sorbonne Univ., Paris, France), Jacques Marchal (IJLRA, Sorbonne Univ., SAINT-CYR-L'ECOLE, France), and Stefan Catheline (LabTAU, INSERM, Lyon, France)

The sensitivity of odontocetes to changes in sound source azimuth and, in particular, elevation (minimum audible angles) has been reported to be superior to those of humans and bats. It has been suggested that binaural/spectral cues might be insufficient to account for this skill. We investigate bone-conducted sound in a short-beaked common dolphin's mandible and attempt to determine whether and to what extent it could contribute to the task of localizing a sound source. Experiments are conducted in a water tank by deploying, on the horizontal and median planes of the skull, sound sources that emit synthetic clicks between 45 and 55 kHz. Elastic waves propagating through the mandible are measured at the pan bones and used to localize source positions via binaural cues, as well as a correlation-based full-waveform algorithm. We find that by making use of the full waveforms, and, most importantly, of their reverberated coda, the accuracy of source localization in the vertical plane can be enhanced. While further experimental work is needed to substantiate this speculation, our results suggest that the auditory system of dolphins might be able to localize sound sources by analyzing the coda of biosonar echoes.

3:55

2pAB10. What bats may do better: Emission beampattern complexity.
Hui Ma, Yanan Zhao (Shandong University - Virginia Tech Int., Lab., Shandong Univ., Jinan, Shandong, China), Liujun Zhang (Mech. Eng., Virginia Tech, Jinan, China), and Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Horseshoe bats (family Rhinolophidae) and the related Old World round-leaf nosed bats (Hipposideridae) have conspicuous emission baffles ("noseleaves") that not only have a high degree of geometric complexity but can also change their shapes under active muscular actuation during emission of the biosonar pulses. Our work has shown that these shape changes occur in tight synchrony with pulse emission and are large enough to affect the acoustic diffraction process. Furthermore, our experiments with bats positioned on an experimental platform have shown that these noseleaf motions impart dynamic signatures onto the emitted ultrasonic wave packets. We also have collected pilot data from flying bats executing natural maneuvers that indicate geometric complexity as well as time-variance in the beampatterns based on a comparison with a loudspeaker that was placed in the same position as the bat to serve as a static reference. An information-theoretic analysis has shown that this emitter dynamics results in the encoding of additional sensory information that is useful, e.g., for direction finding, but future work is needed to determine if and how these effects may fit into the animals' natural biosonar behaviors. It remains to be seen if dolphins are make use of a similar biosonar emission dynamics.

4:10

2pAB11. Ultra high definition 3D tracking of biosonar of Amazon River dolphins and others mammals into the wild.
Gerald Blakefield (MADC EADM, Seattle, Washington), Marie Trone (Valencia Coll., Kissimee, FL), Valentin Barchasz (LIS, CNRS, AMU, Univ Toulon, Toulon, France), Valentin Gies (IM2NP, CNRS, Univ Toulon, Toulon, France), Dave Bonnett (MADC EADM, Seattle, Washington), and Herve Glotin (LIS, CNRS, AMU, Univ. Toulon, USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr)

We designed and built a high capacity digital analog converter : JASON Qualilife DAQ [1] which is capable of sampling at 1 MHz on 5 channels simultaneously to assess cetacean biosonar and the source position and orientation relative to the array using time delay of arrivals of the biosonar acoustical signals. We used JASON with underwater portable arrays housing between 3 and 7 hydrophones (for more than 4 hydrophones we connected two cards sharing one common hydrophone). This has been used to monitor Physeter m. biosonar in near field, and Inia. g. in the Peruvian Amazon from 2014 to 2018. Results demonstrate the efficiency of this advanced low cost scientific instrumentation : it allows with only 4 hydrophones to track in 3D the precise movements of the *I. g.* dolphin, and to determine key behavioral features, like the velocity of its rostrum rotation, while following its highly defined biosonar emissions. This is the first, at our knowledge, that wild amazon dolphin biosonar is described inside its ecosystem. Perspectives on automatic low power trigger [2] for counting of the dolphins and extraction of individual acoustic invariant are given, as well as research avenues on biosonar of other species. *[1] Gies et al, JASON Qualilife Daq, in DCLDE2018, smiot.univ-tln.fr,sabiod.org/bib [2] Fourniol et al., Low-Power Frequency Trigger for Environmental Internet of Things, in IEEE/ASME Mechatronic & Embedded Systems* http://sabiod.org/bib*, 2018 Samples and supp. material: http://sabiod.org/JASON*.

4:25

2pAB12. The micromechanics and bioacoustic behaviour of *Bunaea alcinoe* moth scales.
Zhiyuan Shen, Thomas R. Neil, Daniel Robert (School of Biological Sci., Univ. of Bristol, Office 2B09, Life Sci. Bldg., 24 Tyndall Ave., Bristol BS81TQ, United Kingdom, shenyuan675603@gmail.com), Bruce W. Drinkwater (Dept. of Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), and Marc W. Holderied (School of Biological Sci., Univ. of Bristol, Bristol, United Kingdom)

Through the 65 million year acoustic arms race between moths and bats, different moth species have evolved different defence strategies against bat echolocation. For non-toxic moth species without hearing capability, passive acoustic camouflage is thought to be the most efficient way to evade bat predation. Being the elementary building blocks covering moth wing surfaces, scales have been hypothesized as the main organ creating such acoustic camouflage. There is, however, no understanding for the relation between scale microstructure and wing acoustic performance. This report represents the first effort to numerically and experimentally characterize moth scale biomechanics and vibrational behaviour. 3D microstructures of *Bunaea alcinoe* moth scales have been characterized using various microscopies. A parameterized finite element model has been built to replicate the double-layered perforated scale bio-nanomaterial. Both experimental and numerical analyses have proved that the first three resonance frequencies of a single scale lie within the bat echolocation frequency range. Here, we propose numerical models that explain how the resonances can contribute to the acoustic performance of wings. This study contributes to the on-going discussion of the evolution of ultrasonic camouflage in moths. We aim to use our findings to generate biomimetic light-weight noise mitigation materials.

4:40

2pAB13. Dolphin and bat sonar: Nonlinear acoustic effects.
Thomas G. Muir, Jack A. Shooter, and Mark F. Hamilton (Appl. Res. Laboratories, Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Both dolphin and bat sonars operate in an amplitude and frequency regime where nonlinear effects can be observed, particularly distortion and the generation of harmonics. Computations illustrating these nonlinear

effects are presented, quantifying effects that could be observed in experiments on echolocating signals conducted with wideband sensors. The mathematical model used is the Khokhlov-Zabolotskaya-Kuznetsov (KZK) nonlinear parabolic wave equation, which includes diffraction and absorption for directional beams, with computations performed in the time domain using a finite-difference scheme. Inputs to the computations are sample dolphin echolocation transients (clicks) and bat echolocation tone bursts (FM Chirps—fundamental component), taken from the literature for a few species of high-amplitude emitters. Calculations begin in the water or air media. Propagating high-amplitude dolphin clicks are shown to distort, create harmonics, and suffer nonlinear attenuation at ranges of tens of meters. Propagating high-amplitude bat chirps are shown capable of undergoing some nonlinear distortion at ranges within 10 m. The results may be useful in understanding the significance of nonlinear effects for work with these animals. (Work supported by ARL:UT Austin.)

TUESDAY AFTERNOON, 6 NOVEMBER 2018 ESQUIMALT (VCC), 1:00 P.M. TO 3:40 P.M.

Session 2pAO

Acoustical Oceanography, Animal Bioacoustics, Underwater Acoustics, and Signal Processing in Acoustics: Machine Learning and Data Science Approaches in Ocean Acoustics II

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Shima Abadi, Cochair
University of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011

Invited Papers

1:00

2pAO1. Orcasound app: An open-source solution for streaming live ocean sound to citizen scientists and cloud-based algorithms. Scott Veirs (Beam Reach (SPC), 7044 17th Ave. NE, Seattle, WA 98115, sveirs@gmail.com), Val Veirs (Beam Reach (SPC), Friday Harbor, WA), Paul Cretu (Jotengine, San Francisco, CA), Steve Hicks (Einhorn Eng., Encinitas, CA), and Skander Mzali (Jotengine, San Francisco, CA)

A growing number of hydrophones stream ocean sound data in near-real-time to automated detectors. Many of these streams are also provided to the public, but none yet appeal for citizen scientists to provide real-time detection or classification services. Within the Orcasound hydrophone network—orcasound.net (WA, USA)—live human listeners have always been in friendly competition with real-time automated detectors. To facilitate the participation of citizen scientists in studies of signals from southern resident killer whales and other soniferous organisms of the Salish Sea, in 2017–2018, we developed a low-cost combination of hardware and open-source software to stream audio data from our hydrophones to the headphones of our listeners. The data are archived at and stream from AWS/S3 in real-time (<60 seconds latency) to a browser-based player that is platform- and device-independent. This sets the stage for human detection of signals—mainly the calls, whistles, or clicks of fish-eating orcas—but also novel signals for which we do not yet have classifiers. We will demonstrate the Orcasound app and explain how it interacts synergistically with automated detectors and classifiers, as well as crowdsourced citizen science projects.

1:20

2pAO2. Deep learning for ethoacoustical mapping: Application to a single Cachalot long term recording on joint observatories in Vancouver Island. Herve Glotin (LIS CNRS Univ. Toulon, USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr), Paul Spong, Helena Symonds (Orcalab, Alert Bay, BC, Canada), Vincent Roger (LIS CNRS Univ. Toulon, Toulon, France), Randall Balestriero (Rice Univ., Houston, TX), Maxence Ferrari, Marion Poupard (LIS CNRS Univ. Toulon, Toulon, France), Jared Towers (Fisheries & Oceans Canada, Alert Bay, BC, Canada), Scott Veirs (Orcasound.net, Seattle, Washington), Ricard Marxer, Pascale Giraudet (LIS CNRS Univ. Toulon, Toulon, France), james pilkinton (Fisheries & Oceans Canada, Victoria, BC, Canada), Val Veirs (Orcasound.net, Friday Harbor, Washington), Jason Wood (SMRU, Friday Harbor, Washington), John Ford (Fisheries & Oceans Canada, Victoria, BC, Canada), and Thomas Dakin (Univ. of Victoria, Victoria, BC, Canada)

During February and March, 2018, a lone sperm whale known as Yukusam was recorded first by Orcalab in Johnstone Strait and subsequently on multiple hydrophones within the Salish Sea [1]. We learn and denoise these multichannel clicks trains with AutoEncoders Convolutional Neural Net (CNN). Then, we build a map of the echolocations to elucidate variations in the acoustic behavior of this unique animal over time, in different environments and distinct levels of boat noise. If CNN approximates an optimal kernel decomposition, it requires large amounts of data. Via spline functionals we offer analytics kernels with learnable coefficients do reduce it. We [1-3] identify the analytical mother wavelet to represent the input signal to directly learn the wavelet support from scratch by gradient descend on the parameters of cubic splines [2]. Supplemental material http://sabiod.org/yukusam [1] Balestriero, Roger, Glotin, Baraniuk, Semi-

Supervised Learning via New Deep Network Inversion, arXiv preprint arXiv:1711.04313, 2017 [2] Balestriero, Cosentino, Glotin, Baraniuk, WaveletNet : Spline Filters for End-to-End Deep Learning, Int. Conf. on MachineLearning, ICML, Stockholm, http://sabiod.org/bib, 2018 [3] Spong P., Symonds H., *et al*., Joint Observatories Following a Single male Cachalot during 12 weeks—The Yukusam story, ASA 2018.

1:40

2pAO3. Applying machine-learning based source separation techniques in the analysis of marine soundscapes. Tzu-Hao Lin (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., 2-15, Natsushima, Yokosuka, Kanagawa 237-0061, Japan, schonkopf@gmail.com), Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan), Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan), and Katsunori Fujikura (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., Yokosuka, Japan)

Long-term monitoring of underwater soundscapes provides us a large number of acoustic recordings to study a marine ecosystem. Characteristics of a marine ecosystem, such as the habitat quality, composition of marine fauna, and the level of human interference, may be analyzed using information relevant to environmental sound, biological sound, and anthropogenic sound. Supervised source separation techniques have been widely employed in speech and music separation tasks, but it may not be practical for the analysis of marine soundscapes due to the lack of a database that includes a large mount of paired pure and mixed signals. Even when the paired data is not available, different sound sources with unique spectral or temporal patterns may still be separated by apply semi-supervised or unsupervised learning algorithms. In this presentation, supervised and unsupervised source separation techniques will be demonstrated on long-term spectrograms of a marine soundscape. Separation performances under different levels of simultaneous source influence will also be discussed. In the future, more advanced techniques of source separation are necessary to facilitate the soundscape-based marine ecosystem sensing. An open database of marine soundscape will promote the development of machine learning-based source separation. Therefore, we will open acoustic data tested in this presentation on the Asian Soundscape to encourage the open science of marine soundscape.

2:00

2pAO4. Identification of fish species in estuaries and rivers using recorded soundscape with supervised machine learning. Mohsen Badiey and Javier Garcia-Frias (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu)

Identification of various fish species in estuarine and riverine environments is an important ecological problem at the intersection of various disciplines, including underwater acoustics, signal processing, and fisheries. Advancements in this field can be made by applying machine learning and deep learning technologies. From the vast amount of existing work on machine learning techniques, of special relevance to this problem are those aiming at the detection and labeling of speakers on overlapping speech, where the existing literature is much more reduced. In this paper, we present a dataset collected with a passive recorder placed on a tripod in Delaware Bay estuary during summer 2013 to continuously measure environmental soundscapes for several weeks. By using a library of mid-Atlantic region fish sounds as prior knowledge, we adapt the aforementioned techniques to the detection and identification of various recorded sounds. This adaptation is challenging, as the statistical properties of fish sounds are not the same as those of speech, and requires careful research on how to perform data pre-processing, feature extraction, and model development. The application of the proposed machine learning techniques to future observations will enhance the potential for long-term autonomous ecological studies in bays and rivers.

2:20–2:35 Break

Contributed Papers

2:35

2pAO5. Temporal patterns in Pacific white-sided dolphin communication at Barkley Canyon, with implications for multiple populations. Kristen S. Kanes, Stan E. Dosso (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, kristen.kanes@gmail.com), Tania L. Insua (Ocean Networks Canada, Victoria, BC, Canada), and Xavier Mouy (JASCO Appl. Sci., Victoria, BC, Canada)

Acoustic data collection is a cost-effective approach to evaluating activity patterns of otherwise challenging-to-study offshore cetaceans. However, manual analysis of acoustic data is time-consuming and impractical for large data sets. This study evaluates diel and seasonal patterns in Pacific white-sided dolphin communication through automated analysis of 20 months of continuous acoustic data collected from the Barkley Canyon node of Ocean Networks Canada's NEPTUNE observatory, offshore of Vancouver Island, British Columbia, Canada. In this study, cetacean signals are manually annotated in a sub-set of the data, and 94 time and frequency features of these and other sounds are extracted and used to train random forest classifiers targeting Pacific white-sided dolphin pulsed calls. The performance of binary and multiclass models with various forest sizes, minimum leaf sizes, and confidence thresholds for Pacific white-sided dolphin classification are compared through nested 10-fold cross-validation to select the best model. Vocalizations are classified with the resultant classifier, manually verified, and examined for seasonal and diel patterns. Pacific white-sided dolphins are shown to be vocally active during dawn and day in spring and summer and at dusk and night year-round with reduced overall activity in fall and winter, suggesting both consistent nocturnal and migratory diurnal populations.

2:50

2pAO6. Towards the topology of autoencoder of calls versus clicks of marine mammal. Vincent Roger, Maxence Ferrari, Ricard Marxer (Univ. Toulon, AMU, CNRS, LIS, Toulon, France), Faicel Chamroukhi (LMNO UMR CNRS 6139, Caen, France), and Herve Glotin (Univ. Toulon, AMU, CNRS, LIS, USTV, Ave. Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr)

The goal is to learn the features and the representation adapted for cetacean sound dynamics without any priors. Thus, we develop data driven model to generate voicing and click of cetaceans audio signals. We learn representation and features of stationary or nonstationary emission using neural network from raw audio. We use different types of convolutions (causal, with strides, with dilation [1]), or gradient inversion [2]. Experiments are conducted on various kind of calls of humpback whales from nips4b challenge [3] or Orca whale. We compare the topology for transient encoding on Physeters and Inia g. For each model, we detail the resulting

filters and discuss on the topology. We acknowledge Region PACA and NortekMED for Roger's Phd grant, & DGA and Région Haut de France for Ferrari's Phd grant. [1] Oord, Dieleman, Zen, Simonyan, Vinyals, Graves *et al.* Wavenet : A generative model for raw audio, arXiv:1609.03499, 2016 [2] Balestriero, Roger, Glotin, Baraniuk, Semi-Supervised Learning via New Deep Network Inversion, arXiv:1711.04313, 2017 [3] Glotin, LeCun, Mallat *et al.* Proc. 1st wkp on Neural Information Processing for Bioacoustics NIPS4B, joint to NIPS Alberta USA, 2013 http://sabiod.org/nips4b/challenge2.html, http://sabiod.org/NIPS4B2013_book.pdf

3:05

2pAO7. PyEcholab: An open-source, python-based toolkit to analyze water-column echosounder data. Carrie C. Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), Rick Towler (Alaska Fisheries Sci. Ctr., NOAA NMFS, Seattle, WA), Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Boulder, CO), Randy Cutter (Southwest Fisheries Sci. Ctr., NOAA NMFS, La Jolla, CA), and J. Michael Jech (Northeast Fisheries Sci. Ctr., NOAA NMFS, Woods Hole, MA)

Water-column echosounder data are widely used for a diversity of research objectives from fisheries abundance to identifying methane seeps to characterization of the seafloor. However, these data are voluminous, complex, and recorded in instrument-specific binary file formats. Tools to process these data are limited to a few commercial applications or custom programs developed by researchers, leaving potential users without sufficient financial resources or programming knowledge to use the data. To address this problem, NOAA Fisheries, University of Colorado Boulder's Cooperative Institute for Research in Environmental Sciences, and NOAA National Centers for Environmental Information (NCEI) scientists are developing PyEcholab. PyEcholab is an open-source, python-based system for reading, processing, and visualizing water-column echosounder files. Currently, the system is being developed to meet existing NCEI processing and visualization needs but PyEcholab's base classes and open architecture provide a framework for developing new file readers, processing algorithms, and visualization techniques that can be modularized into the system. The long-term vision is to engage the community in an open-source effort that continually grows PyEcholab's capabilities and expands the accessibility of water-column echosounder data.

Invited Paper

3:20

2pAO8. Toward scalable, reproducible, and open ocean acoustic research. Valentina Staneva, Amanda Tan (eSci. Inst., UW-IT, Univ. of Washington, Campus Box 351570, 3910 15th Ave. NE, Seattle, WA 98105, vms16@uw.edu), Divya Panicker (Biological Oceanogr., Univ. of Washington, Seattle, WA), and Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

As our ability to collect long-term acoustical signals grows we face the challenges of analyzing large datasets, ensuring our findings are reproducible, and sharing our work with collaborators. In this talk, we will give an overview of frameworks and tools that can facilitate those tasks. We will show how the Jupyter Notebook environment serves as a hub for quick prototyping, literate programming, interactive analysis and visualization, and a gateway to cloud computing. We will provide examples of processing of larger-than-memory ocean acoustic recordings from the comfort of our own laptop. We will further outline best practices for sharing code and data, and discuss the needs and approaches to achieve scalable, reproducible, and open acoustic research.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 SIDNEY (VCC), 1:00 P.M. TO 3:45 P.M.

Session 2pBAa

Biomedical Acoustics and Physical Acoustics: Shock Waves and Ultrasound for Calculus Fragmentation

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Pennsylvania State University, Penn State, 201E Applied Sciences Building, University Park, PA 16802

Michael R. Bailey, Cochair
Applied Physics Lab, University of Washington, Center for Industrial and Medical Ultrasound, APL-UW, Seattle, WA 98105

Contributed Papers

1:00

2pBAa1. Update on clinical trials results of kidney stone repositioning and preclinical results of stone breaking with one ultrasound system. Michael R. Bailey, Yak-Nam Wang, Wayne Kreider, Bryan W. Cunitz (Appl. Phys. Lab, Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, APL-UW, Seattle, WA 98105, mbailey@uw.edu), Jessica Dai, Jonathan Harper, Helena Chang, Matthew D. Sorensen (Dept. of Urology, Univ. of Washington, Seattle, WA), Ziyue Liu (Dept. of Biostatistics, Indiana University-Purdue Univ. Indianapolis, Indianapolis, IN), Oren Levy (SonoMotion, Inc., San Francisco, CA), Barbrina Dunmire (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA)

Our goal is an office-based, handheld ultrasound system to target, detach, break, and/or expel stones and stone fragments from the urinary space to facilitate natural clearance. Repositioning of stones in humans (maximum 2.5 MPa, and 3-second bursts) and breaking of stones in a porcine model (maximum 50 cycles, 20 Hz repetition, 30 minutes, and 7 MPa peak negative pressure) have been demonstrated using the same 350-kHz probe. Repositioning in humans was conducted during surgery with a ureteroscope in the kidney to film stone movement. Independent video review confirmed stone movements ($\geq$ 3 mm) in 14 of 15 kidneys (93%). No serious or unanticipated adverse events were reported. Experiments of burst wave lithotripsy (BWL) effectiveness on breaking human stones implanted in the porcine bladder and kidney demonstrated fragmentation of 4 of 4 stones on post mortem dissection. All clinical pathology, hematology, and urinalysis for a 1-week survival study with the BWL exposures in 10 specific pathogen free pigs were within normal limits. These results demonstrate that repositioning of stones with ultrasonic propulsion and breaking of stones with BWL are safe and effective. [Work supported by NIH P01 DK043881, K01 DK104854, and R44 DK109779.]

1:15

2pBAa2. Acoustic radiation force acting on a spherical scatterer in water: Measurements and simulation. Maria M. Karzova, Anastasiya V. Nikolaeva, Sergey Tsysar (Phys. Faculty, Moscow State Univ., Leninskie Gory 1/2, Phys. Faculty, Dept. of Acoust., Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Vera Khokhlova, and Oleg Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, USA; Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Acoustic radiation force (ARF) is successfully used in a recently proposed ultrasonic technology for kidney stone propulsion. The planning of the treatment requires calibration of ARF for stones of different dimensions and locations. However, such calibration remains a problem. Here, a method of measuring ARF acting on a mm-sized spherical object positioned on the axis of a focused ultrasound beam is proposed and tested. Acoustic field was generated by a single-element 1.072 MHz transducer of 100 mm aperture and 70 mm focal length positioned at the bottom of the water tank. Measurements were performed for nylon and glass spherical scatterers with diameters from 2 to 4 mm located at different distances from the source along the vertical beam axis. For each scatterer, the source power was gradually decreased until the scatterer started to fall down from the trap. At this threshold power of the source, the value of ARF was determined as a difference between the gravity and buoyancy forces acting on the scatterer. At other source power outputs, the value of ARF was linearly scaled. ARF was also calculated from pressure distributions reconstructed from acoustic hologram of the source and physical parameters of the scatterers. Experimental and theoretical results were found in a good agreement. It was shown that the most effective pushing was observed at distances where the beam was slightly wider than the scatterer. [Work supported by the stipend of the President of Russia (SP-2621.2016.4), RFBR №18-32-00659, and NIH P01 DK43881.]

1:30

2pBAa3. Generation of guided waves during burst wave lithotripsy as a mechanism of stone fracture. Adam D. Maxwell (Urology, Univ. of Washington School of Medicine, 1013 NE 40th st, Seattle, WA 98105, amax38@u.washington.edu), Brian MacConaghy, Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation)

Burst wave lithotripsy (BWL) is an experimental method to noninvasively fragment urinary stones by short pulses of focused ultrasound. We are investigating physical mechanisms of stone fracture during BWL to better optimize this procedure. In this study, we used photoelasticity imaging as a method to visualize elastic wave dynamics in model stones. Epoxy and glass stone models were made into cylindrical, rectangular, or irregular geometries and exposed in a degassed water bath to focused ultrasound bursts at different frequencies. A high-speed camera was used to record images of the stone during exposure through a circular polariscope backlit by a monochromic flash source. Results showed development of periodic stresses in the stone body with a pattern dependent on frequency. These were identified as guided wave modes in cylinders and plates, which formed standing waves upon reflection from the distal surfaces of the stone model, causing periodic stress positions. Measured phase velocities compared favorably to specific numerically calculated modes dependent on frequency and material. Artificial stones exposed to BWL produced cracks at positions anticipated by this mechanism. These results support guided wave production and reflection as a mechanism of stone fracture in BWL. [Work supported by K01 DK104854 and P01 DK043881.]

1:45

2pBAa4. Impact of stone characteristics on cavitation in burst wave lithotripsy. Christopher Hunter (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 ne 40th St., Seattle, WA 98105, chunter6@uw.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan Cunitz, Barbrina Dunmire (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), James C. Williams (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), Michael Bailey, Akshay Randad, and Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Non-invasive kidney stone treatments such as shock wave lithotripsy (SWL) and burst wave lithotripsy (BWL) rely on the delivery of pressure waves through tissue to the stone. In both SWL and BWL, the potential to hinder comminution by exciting cavitation proximal to the stone has been reported. To elucidate how different stones alter prefocal cavitation in BWL, different natural and synthetic stones were treated *in vitro* using a therapy transducer operating at 350 kHz (peak negative pressure 7 MPa, pulse length 20 cycles, pulse repetition frequency 10 Hz). Stones were held in a confined volume of water designed to mimic the geometry of a kidney calyx, with the water filtered and degassed to maintain conditions for which the cavitation threshold (in the absence of a stone) matches that from *in vivo* observations. Stone targeting and cavitation monitoring were performed via ultrasound imaging using a diagnostic probe aligned coaxially with the therapy transducer. Quantitative differences in the extent and location of cavitation activity were observed for different stone types—e.g., "softer" stones (natural and synthetic) that disintegrate into dusty fragments produced larger prefocal cavitation clouds. Future work will focus on correlation of such cavitation metrics with stone fragmentation. [Funding support by NIH P01-DK043881, K01-DK104854.]

Invited Paper

2:00

2pBAa5. Modeling and numerical simulation of the bubble cloud dynamics in an ultrasound field for burst wave lithotripsy.
Kazuki Maeda (Univ. of Washington, Seattle, WA), Tim Colonius (California Inst. of Technol., 1200 E. California Blvd., Pasadena, CA 91125, colonius@caltech.edu), Adam D. Maxwell, Wayne Kreider, and Michael R. Bailey (Univ. of Washington, Seattle, WA)

Modeling and numerical simulation of bubble clouds induced by intense ultrasound waves are conducted to quantify the effect of cloud cavitation on burst wave lithotripsy, a proposed non-invasive alternative to shock wave lithotripsy that uses pulses of ultrasound with an amplitude of $O(1)$ MPa and a frequency of $O(100)$ kHz. A unidirectional acoustic source model and an Eulerian-Lagrangian method are developed for simulation of ultrasound generation from a multi-element array transducer and cavitation bubbles, respectively. Parametric simulations of the spherical bubble cloud dynamics reveal a new scaling parameter that dictates both the structure of the bubble cloud and the amplitude of the far-field, bubble-scattered acoustics. The simulation further shows that a thin layer of bubble clouds nucleated near a kidney stone model can shield up to 90% of the incoming wave energy, indicating a potential loss of efficacy during the treatment due to cavitation. Strong correlations are identified between the far-field, bubble-scattered acoustics and the magnitude of the shielding, which could be used for ultrasound monitoring of cavitation during treatments. The simulations are validated by companion experiments *in* vitro. [The work was supported by NIH P01-DK043881 and K01-DK104854.]

Contributed Papers

2:15

2pBAa6. Burst wave lithotripsy: An *in vivo* demonstration of efficacy and acute safety using a porcine model. Yak-Nam Wang, Wayne Kreider, Christopher Hunter, Bryan Cunitz, Jeffrey Thiel, Frank Starr, Jessica Dai, Yasser Nazari, Donghoon Lee (Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, ynwang@uw.edu), James C. Williams (Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), Michael R. Bailey, and Adam D. Maxwell (Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a new non-invasive method for stone comminution using bursts of sub-megahertz ultrasound. A porcine model of urolithiasis and techniques to implement BWL treatment have been developed to evaluate its effectiveness and acute safety. Five human calcium oxalate monohydrate stones (6–7 mm) were hydrated, weighed, and surgically implanted into the kidneys of three pigs. Transcutaneous stone treatments were performed with a BWL transducer coupled to the skin via an external water bath. Stone targeting and treatment monitoring were performed with a co-aligned ultrasound imaging probe. Treatment exposures were applied in three 10-minute intervals for each stone. If sustained cavitation in the parenchyma was observed by ultrasound imaging feedback, treatment was paused and the pressure amplitude was decreased for the remaining time. Peak negative focal pressures between 6.5 and 7 MPa were applied for all treatments. After treatment, stone fragments were removed from the kidneys. At least 50% of each stone was reduced to <2 mm fragments. 100% of four stones were reduced to <4 mm fragments. Magnetic resonance imaging showed minimal injury to the functional renal volume. This study demonstrated that BWL can be used to effectively fragment kidney stones with minimal injury. [Funding support: NIH P01-DK043881, NIH K01-DK104854.]

2:30–2:45 Break

2:45

2pBAa7. Design of a transducer for fragmenting large kidney stones using burst wave lithotripsy. Akshay P. Randad (Mech. Eng. Dept., Univ. of Washington, Stevens Way, Box 352600, Seattle, WA 98195, aprandad@u.washington.edu), Mohamed A. Ghanem (Dept. of Aeronautics and Astronautics, Univ. of Washington, Seattle, WA), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, APL, Univ. of Washington, Seattle, WA), and Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a potential noninvasive treatment for breaking kidney stones. BWL requirements of high-pressure output, limited aperture for acoustic window, and specific focal length and frequency constrain the focal beam width. However, BWL is most effective only on stones smaller than the beam width. We tested a porous piezoelectric material (PZ36) to increase the output power and designed acoustic lenses that broaden the beam. A weighted iterative angular spectrum approach was used to calculate the source phase distribution needed to generate desired cross sectional focal beam profiles each of 12 mm width. The phase calculations were then 3D printed as holographic lenses placed over a circular aperture of 80-mm diameter, 350 kHz PZ36 to produce the desired beam at 85 mm depth. The difference in simulated beam width and that measured by hydrophone was <1 mm, and the structural-similarity index value was greater than 0.65. The differences in structures were due not to shape and size of the 6-dB contours but to amplitude distribution within the contour. In conclusion, this design approach combined with 3D printing provide a way to tailor focal beam profiles for lithotripsy transducers. [The work was supported by NIDDK P01-DK043881 and K01-DK104854.]

3:00

2pBAa8. Acoustic bubble coalescence and dispersion for enhanced shockwave lithotripsy. Hedieh Alavi Tamaddoni and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, 2135 Carl A. Gerstacker Bldg., 2200 Bonisteel Blvd., Ann Arbor, MI 48109, alavi@umich.edu)

The goal of this study is to develop methods to improve shockwave lithotripsy (SWL). Although cavitation on the surface of a stone may aid fragmentation, cavitation away from the stone may block subsequent shockwaves or cause tissue damage. In previous work, we have shown low amplitude acoustic bursts immediately *after* each shockwave can force cavitation bubbles to coalesce enhancing SWL efficacy. In this study we examined the feasibility of acoustically dispersing bubbles away from the propagation path immediately *before* the arrival of the next shockwave. A clinical Dornier lithotripter was used with an in-house made transducer to generate Acoustic Bubble Coalescence and Dispersion (ABCD) pulses fired at different timing with respect to each shockwave. Model stones were treated with 2500 shockwaves at 30 shocks/min or 120 shocks/min and four different ABCD pulse sequences *in-vitro*. Results showed fragments larger than 2 mm were significantly reduced for all four ABCD sequences cases at 120 shocks/min. The remnant mass of fragments larger than 2 mm was 0.16% at low rate of 30 shocks/min, and 15.81% at 120 shocks/min without ABCD, and 0.19% at 120 shocks/min with ABCD. These results suggest that dispersing residual bubbles can aid in fragmentation efficiency allowing faster SWL treatments.

3:15

2pBAa9. Confocal lens focused piezoelectric lithotripter. Gilles Thomas, Jean-Yves Chapelon, Alain Birer, and Cyril Lafon (U1032, INSERM, 151, cours Albert Thomas, Lyon 69424, France, gilles.thomas@inserm.fr)

Usually, piezoelectric lithotripters consist of a high quantity of small flat piezoelectric discs tied to a spherical structure, where the focus is the

geometric center of the sphere. The small diameter of the ceramics compared to the distance to the focus of the lithotripter means that only a fraction of the surface pressure will arrive to the focus, and also results in a very small focal diameter. This work focus on the evaluation of a new type of piezoelectric lithotripter with similar dimensions of a commercial lithotripter and composed of either 3 or 4 large lens focused piezoelectric transducers with focal pressure up to 25 MPa each, set either in a confocal C-shape or confocal ring-shape. Each transducer is made with a 92 mm diameter flat piezoelectric ceramic disc of 220, 300, or 400 kHz thickness frequency and the acoustic lens shape was calculated using finite element optimization in order to maximize its focusing capability. Comparison of artificial stone comminution efficiency depending on the frequency, pressure, and the setup of the piezoelectric transducers were made and compared to commercially available lithotripters. [Work supported by an industrial grant from EDAP-TMS.]

3:30

2pBAa10. Experimental observations and numerical modeling of lipid-shell microbubbles with stone targeting moieties for minimally-invasive treatment of urinary stones. Yuri A. Pishchalnikov, William Behnke-Parks (Applaud Medical Inc., 953 Indiana St., San Francisco, CA 94107, yurapish@gmail.com), Kazuki Maeda (Dept. of Mech. Eng., Univ. of Washington, Seattle, WA), Tim Colonius (Dept. of Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Matt Mellema, Matt Hopcroft, Alice Luong (Applaud Medical Inc., San Francisco, CA), Scott Wiener, Marshall Stoller (Dept. of Urology, Univ. of California, San Francisco, CA), Thomas Kenny (Dept. of Mech. Eng., Stanford Univ., Stanford, CA), and Daniel Laser (Applaud Medical Inc., San Francisco, CA)

Products incorporating stone-targeting microbubbles have recently entered human clinical trials as a new minimally-invasive approach to treat urinary stones. Lipid-shell, gas-core microbubbles can be introduced into the urinary tract through a catheter. Calcium-binding moieties incorporated into the lipid shell can facilitate binding to stones. The microbubbles can be excited by an extracorporeal source of low-intensity ultrasound. Alternatively, the microbubbles can be excited by an intraluminal source, such as a fiber-optic laser. With either excitation technique, stone-targeting microbubbles can significantly increase rates of erosion, pitting, and fragmentation of stones, as has recently been reported for *in-vitro* experiments with synthetic stones [Wiener *et al*., J. Urology, v.199, no.4S, e322 (2018)]. We report here on new experiments using high-speed photography to characterize microbubbles expansion of cracks within a stone and resultant breaking-off of stone fragments. Numerical modeling shows that the direction of micro-jets produced by collapsing stone-bound microbubbles depends strongly on bubble shape and stand-off distance. For a wide range of stand-off distances and bubble shapes, microbubble collapse is associated with pressure increases of some two orders of magnitude compared to the excitation source pressures. This *in-vitro* study provides key insights into the use of stone-targeting microbubbles in treatment of urinary stones.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 — COLWOOD 1/2 (VCC), 1:00 P.M. TO 5:10 P.M.

Session 2pBAb

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

Invited Papers

1:00

2pBAb1. Understanding the influence of grain scattering noise on array imaging and defect characterisation. Bruce W. Drinkwater, Long Bai, and Alexander Velichko (Mech. Eng., Univ. of Bristol, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

The performance of ultrasonic inspection degrades for coarse grained materials because of the multiple scattering effects caused by the material microstructure. The influence of grain scattering noise on detection and characterisation of defects is studied in this paper. Two parameters, mean grain size and grain size variation, are used to describe a particular grain structure, which is included in a finite element (FE) model to simulate the array data of a defect with the presence of grain noise. The ultrasonic attenuation and multiple scattering effects become more significant at high frequencies, and so a low frequency is shown to be preferable for defect detection. However, the frequency should also be higher than a certain limit for the defect to be characterisable based on its scattering matrix. Here we extract the noise scattering matrix due to the grain scattering from the simulated array data and use its statistical distribution is used to determine the characterisation uncertainty. Preliminary results obtained at 2.5 MHz for stainless-steel show that 1 mm (i.e., 0.45λ) cracks can be sized reliably when the mean grain size is less than 0.2 mm. However, when the mean grain size exceeds 0.15 mm, it is difficult to distinguish the 1 mm cracks from volumetric defects of the same size.

1:20

2pBAb2. Ultrasonic characterization of synthetic three-dimensional polycrystals. Musa Norouzian, Nathanial Matz, Showmic Islam, and Joseph A. Turner (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu)

Ultrasonic nondestructive testing has been increasingly used to characterize heterogeneities of polycrystalline materials. With such techniques, the interactions of coherent ultrasonic waves with grain boundaries result in scattering. Such scattered waves carry information regarding the physical properties of the scatterer. Therefore, microstructural information can be obtained by quantifying the scattered response. Current diffuse ultrasonic backscatter models include several assumptions about the macroscopic and microscopic properties of the polycrystals. In this presentation, the sensitivity of grain size characterization to such assumptions is investigated using simulated microstructures. Several polycrystals with cubic crystal symmetry and randomly oriented grains are simulated using Dream.3D. This study applies the single scattering model in which the longitudinal-to-longitudinal configuration is considered for the incident and the scattered waves and limited to the weakly-scattering regime. In each configuration, the theoretical results are compared with results from the synthetic volumes. The results demonstrate distinct differences between the theory and the simulation. For example, the theoretical scattering cross section for a Voigt-averaged copper polycrystal at 15 MHz is found to be about three times larger than the value based on the Dream.3D microstructure. Finally, the influence of grain size distribution and grain elongation on the ultrasonic models are investigated.

1:40

2pBAb3. Finite element modelling of scattering of an acoustic wave by particles in a fluid: Shear and thermal effects. Valerie J. Pinfield, Derek M. Forrester (Chemical Eng. Dept., Loughborough Univ.,Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk), and Jinrui Huang (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough, Leicestershire, United Kingdom)

We investigate, using finite element modelling, the effects of viscous and thermal dissipation in the region around a particle in a liquid subjected to an acoustic field. The linearized thermo-acoustic equations for propagation in a viscous liquid are used in the model. In the Rayleigh scattering regime, with Mega-Hertz frequencies and nano- or micron-sized particles, the shear and thermal decay lengths are of order micrometres and are therefore challenging to resolve accurately using finite element models. We report investigations of the effect of particle size and frequency on the magnitude of the thermal and shear wave fields and their decay length and compare with analytical predictions. We present a determination of the viscous and thermal power dissipation around a single particle, and for small clusters of particles. Here we identify the effect of interaction of the shear and thermal fields with neighbouring particles through mode reconversion, leading to a reduction in dissipation losses when interparticle separation is of the order of the thermal or shear decay length.

Contributed Paper

2:00

2pBAb4. Quantitative assessment of angiogenic microvasculature using Ultrasound multiple scattering with microbubble contrast agents. Kaustav Mohanty (Mech. and Aerosp. Eng., North Carolina State Univ., 3147 B, 911 Oval Dr., College of Eng., EB-3, Raleigh, NC 27606, kmohant@ncsu.edu), Virginie Papadopoulou, Isabel Newsome, Paul A. Dayton (Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

In tumors, angiogenesis (formation of new blood vessels) is established as a biomarker of malignancy. A random, isotropic, high-density vessel network is linked to tumor invasiveness. Ultrasound quantification of angiogenic micro-architecture could help increase the specificity of ultrasound for cancer diagnosis. We exploit multiple scattering by microbubbles populating angiogenic networks to characterize an effective medium diffusivity via the diffusion constant (D). The inter-element response matrix is acquired using an L11-4v linear array transducer operating at 7.8 MHz. D is computed from the time evolution of the incoherent contribution to the backscattered intensity in the near field. D is measured in fibrosarcoma tumors subcutaneously implanted in rats (n = 16), and in control, healthy tissue (n = 18). D is measured for two different orientations (Coronal and Transverse) and the anisotropy of the microvasculature is evaluated via the ratio of the D values obtained in the two different orientations. D was found significantly different between control (1.38 $\pm$ 0.51 mm^2/μs) and tumor (0.65 $\pm$ 0.27 mm^2/μs) (p<0.01) and anisotropy of the angiogenic network was observed in control cases (1.62 $\pm$ 0.911). For further validation, these results were corroborated with vascular density measurements from acoustic angiography data, confirming increased vessel density in tumors compared to controls.

Invited Papers

2:15

2pBAb5. Experimental assessment of guided waves far-field scattering around damage in metallic and composite structures. Patrice Masson, Nicolas Quaegebeur (Mech. Eng. Dept., Universite de Sherbrooke, 2500 Blvd. Universite, Sherbrooke, QC J1K 2R1, Canada, Patrice.Masson@USherbrooke.ca), Mohammad H. Sherafat, Larry Lessard, and Pascal Hubert (Mech. Eng., McGill Univ., Montreal, QC, Canada)

Experimental damage simulation is useful for designing ultrasonic guided-wave based systems for non-destructive evaluation (NDE) and structural health monitoring (SHM). However, simulating the scattering of guided waves with geometrical (rivets, thickness changes, stiffeners, and extrusions) or damage features (fatigue cracks, fillet cracks, delaminations, and disbonds) remains a challenge. The objective of this work is to assess to which extent the interaction of ultrasonic guided waves with typical damage can be captured with an experimental model for a metallic structure and a composite structure. For the metallic structure, real fatigue cracks around a rivet hole are simulated by machined notches, while, for the composite structure, the impact damage is simulated by a single artificial delamination introduced into the laminate using two circular Teflon tapes during manufacturing. This paper implements an experimental methodology for estimating the far-field scattering for both simulated and real damage. Two co-localized rectangular piezoceramics are used to generate the guided waves and non-contact measurement is performed using a three-dimensional laser Doppler vibrometer (3D-LDV) to extract the required information for evaluation of the reflection, transmission, as well as the scattering behavior of the waves. The corresponding coefficients as a function of frequency, incident angle, and type of damage are extracted.

2:35

2pBAb6. Interaction of guided waves with defects in a multilayered cylinder by an asymptotic approach. Aditya Krishna (I2M, CEA, Talence, Gironde, France), Eric Ducasse (I2M Arts et Metiers ParisTech, Talence, Gironde, France), and Marc Deschamps (I2M, Universite Bordeaux, 351 cours de la liberation, Talence, Gironde 33400, France, marc.deschamps@u-bordeaux.fr)

This work deals with the problem of the propagation of an elastodynamic field radiated by a source in a cylindrical layered medium which interacts with and is diffracted by a defect. At low frequencies, where the defect size is much smaller than the wavelength, this interaction can be approximated by a point source, located at the defect position. This secondary source is expressed by the Green function and its derivative. The Green function, which describes the response of an undamaged cylinder, is calculated using the canonical form of the wave equation initially expressed as a function of the spatial and temporal variables. Performing the Laplace in time and Fourier transforms along the cylinder axis, this equation is written as an ordinary differential equation with respect to the radial position. The solution for the transversely isotropic case is obtained by adopting the partial wave formulation, expressed as a combination of the modified Bessel's functions of the first and second kind. Having assembled the layers, numerical inverse transforms are performed to obtain the real wave fields. This technique allows for reduced computational costs and faster calculation times and could be used for the non destructive testing of embedded pipes and tubes.

2:55–3:10 Break

3:10

2pBAb7. Harmonic generation and frequency mixing at nonlinear imperfect interfaces. 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)

Imperfect interfaces can be found in a variety of natural and artificial systems. Some examples of such interfaces that are objects of ultrasonic evaluation and diagnostics include contacting mechanical components as well as closed defects in structural components. A remarkable feature of these interfaces is that they exhibit highly nonlinear responses to ultrasonic waves. Harmonic generation (generation of, e.g., double frequency signals at the incidence of a wave with a certain frequency) and frequency mixing (generation of sum/difference frequency signals at the incidence of waves of two different frequencies) are typical phenomena at these interfaces. In this presentation, a theoretical analysis of these phenomena is presented based on modeling the interface as a nonlinear spring-type interface between two similar linearly elastic media. Unlike the corresponding nonlinear acoustic effects due to material nonlinearity, the interfacial nonlinear effects considered here do not require the spatial accumulation of nonlinearly generated harmonic or sum/difference frequency components. The propagation directions and amplitudes of these nonlinearly generated components are derived for plane mono- or dichromatic incident waves. Some relevant experimental results for a contacting interface of metallic blocks subjected to different applied pressures are also presented.

Contributed Papers

3:30

2pBAb8. Interfacial effects in underwater acoustic panel measurements. Jeffrey Szabo and Adam Bent (Atlantic Res. Ctr., Defence Res. and Development Canada, PO Box 1012, 9 Grove St., Dartmouth, NS B2Y3Z7, Canada, jeff.szabo@drdc-rddc.gc.ca)

Underwater acoustic panel measurements were conducted on single layers of rubber or metal, and on rubber/metal bilayers that had been fabricated using various adhesives or attachment methods. Reflection and transmission measurements were carried out on submerged panels using a parametric array source. It was found that the condition of the various interfaces in the system (water/rubber, water/metal, or rubber/metal) had a dominant effect on the measured acoustic properties. Single layer (metal or rubber) panels with smooth surfaces behaved in accordance with theoretical predictions, but panels with even a small amount of surface roughness did not. This was attributed to a non-ideal water/sample interface, most likely due to the presence of air bubbles. The effect of various surface treatments on the acoustic behaviour of the panels was investigated.

Different adhesion methods for fabricating aluminum/rubber bilayers were tested for their practicality and effects on underwater acoustic properties. Nitrile butadiene rubber and aluminum panels were adhered to one another using pressure sensitive adhesive, contact cement, banded duct tape, or epoxy, and tested in transmission and reflection. The results were strongly dependent on adhesion method, with only one method (epoxy cured under 15 psi pressure) giving results that matched theoretical predictions.

3:45

2pBAb9. Reflection of an ultrasonic wave from the bone-implant interface: A numerical study. Yoann Hériveaux (Laboratoire MSME, CNRS, UPEC - Fac des Sci. - Laboratoire MSME, 61, Ave. du général de Gaulle, Créteil 94010, France, yoann.heriveaux@u-pec.fr), Vu-Hieu Nguyen (Laboratoire MSME, Université Paris-Est, Creteil, France), and Guillaume Haiat (Laboratoire MSME, CNRS, Créteil, France)

Quantitative ultrasound are used to characterize and stimulate osseointegration processes at the bone-implant interface (BII). However, the interaction between an ultrasonic wave and the implant remains poorly understood. This study aims at investigating the sensitivity of the ultrasonic response to the microscopic and macroscopic properties of the BII and to osseointegration processes. The reflection coefficient R of the BII was modeled for different frequencies using a two-dimensional finite element model. The implant surface was modeled by a sinusoidal function with varying amplitude and spatial frequency and then by considering actual implant surface profiles. A soft tissue layer of thickness W was introduced between bone tissue and the implant in order to model non-mineralized fibrous tissue. For microscopic roughness, R is shown to increase from around 0.55 until 0.9 when $k.W$ increases from 0 to 1 and to be constant for $k.W>1$. These results show that R depends on the properties of bone tissue located at a distance comprised between 1 and 25 μm from the implant surface. For macroscopic roughness, R is highly dependent on the roughness amplitude, which may be explained by phase cancellation and multiple scattering effects for high roughness parameters.

4:00

2pBAb10. *In vitro* and *in vivo* comparison between methods based on quantitative ultrasound and on resonance frequency analysis to assess dental implant stability. Romain Vayron (Multiscale Modeling and Simulation Lab., CNRS, Créteil, France), 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)

Resonance frequency analyses (RFA) and quantitative ultrasound (QUS) methods have been suggested to dental assess implant stability. The aim of this study was to compare the results obtained using these two techniques *in vitro* and *in vivo*. Implants were inserted in bone phantoms with different values of density and cortical thickness to assess the effect of bone quality on the ultrasonic indicator (UI) and on the ISQ values. 81 identical implants were inserted in the iliac crests of 11 sheep. The QUS and RFA measurements were realized after different healing times. *ISQ* values increase and *UI* values decrease when i) the bone density and ii) cortical thickness increase. The error realized on the estimation of the trabecular density (respectively cortical thickness) with the QUS device is around 4 (respectively, 8) times lower compared to that made with the RFA technique. The error made on the estimation of the healing time using the QUS technique was 10 times lower than using the RFA technique. The results show that ultrasound technique provides a better estimation of different parameters related to the implant stability compared to the RFA technique, paving the way towards the development of a decision support system to dental surgeons.

Invited Paper

4:15

2pBAb11. Experimental behaviour of an acoustic wave in a medium comprising self-similar (fractal) objects. Elsy Mandelbrot (ENSAV, Versailles, France), Bernard Sapoval (LPMC, Ecole Polytechnique, Palaiseau, France), and Vincent Gibiat (ICA, Toulouse Univ., 118 Rte. de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr)

After the pioneering work of one of the authors (Sapoval *et al.* J. Acoust. Soc. Am. **104**, 1997), the study of acoustic propagation over fractal boundaries came up against the difficulty of experimenting with self-similar objects immersed in two- and three-dimensional space. The idea of one of us to use the capabilities offered by three-dimensional printing has made it possible to overcome this difficulty and produce numerous objects, as small or large as desired, in large numbers. We will present the experimental results obtained for acoustic propagation, reflection coefficient, attenuation and trapping in the case of an half fractal octahedron, isolated as well as in the case of a flat paving consisting of an important number of these objects.

Contributed Paper

4:35

2pBAb12. Phase velocity dispersion estimation of viscoelastic materials by two-point wavelet-based method. Piotr Kijanka (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, kijanka.piotr@mayo.edu), Lukasz Ambrozinski (Dept. of Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Ultrasound shear wave elastography (SWE) is a promising imaging modality used for noninvasive, quantitative evaluation of tissue mechanical properties. This method uses an acoustic radiation force to produce laterally propagating shear waves that can be tracked to obtain the wave velocity. One of the ways to explore the viscoelasticity is through examining the shear wave velocity dispersion curves. In this paper, we present an alternative method to the classical two-dimensional Fourier transform (2D-FT), a two-point wavelet transformation method. We use the Morlet wavelet function as the mother wavelet to filter two shear waves at different locations. We examined how starting distance from the push and distance between the two locations affected the shear wave velocity dispersion. We tested this method on digital phantom data created using local interaction simulation approach (LISA) in viscoelastic media and on data acquired from phantom experiments. We compared results from the two-point method with the 2D-FT technique. The two-point method provided dispersion curves estimation with lower errors over a wider frequency band. Tests conducted showed that the two-point technique gives results with better accuracy in simulation results and can be used to measure phase velocity of viscoelastic materials.

Invited Paper

4:50

2pBAb13. Ultrasonic and X-Ray tomography inspection of a woven glass reinforced composite damaged by fatigue loading. Nada Miqoi (LEM3 - UMR CNRS 7239, ENSAM - Arts et Métiers ParisTech, Metz, France), Pascal Pomarède, Nico F. Declercq (Mech. Eng., UMI Georgia Tech – CNRS 2958, Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France, nico.declercq@me.gatech.edu), Laurent Peltier, and Fodil Meraghni (LEM3 - UMR CNRS 7239, ENSAM - Arts et Métiers ParisTech, Metz, France)

An investigation of a polyamide 6.6/6 composite reinforced with woven glass fibers is described with a focus on its response to fatigue solicitations. The main purpose is to elaborate a quantitative and a qualitative study of the induced damage as well as its detectability. To do so, several nondestructive testing methods were used. The different damage mechanisms, relative to fatigue loading, were identified using X-Ray tomography as a reference for the ultrasonic investigations. As a result, each damage mechanism is visualized and a measure of the induced void is established for comparison. An ultrasonic investigation based on C-scans and guided waves is performed under different signal analysis approaches in an attempt to extract as much information as possible. The discoveries and experiences are important in the framework of a collaboration with the automotive industry en for biomechanical applications.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 SALON A (VCC), 1:00 P.M. TO 3:30 P.M.

Session 2pED

Education in Acoustics: Measuring Educational Outcomes

Andrew A. Piacsek, Chair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Chair's Introduction—1:00

Invited Papers

1:05

2pED1. Measuring students' learning gains with pre/post assessment. John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu) and Kathleen E. Wage (ECE Dept., George Mason Univ., Fairfax, VA)

Pre/post protocols assess students with the same instrument at the start and end of a course to measure changes in their understanding. In contrast, traditional "final exam" assessment measures only the students' understanding but not their progress. Since pre/post testing measures learning gains, it is an essential tool for evaluating the effectiveness of different learning formats, e.g., standard lecture versus active learning. Concept inventories (CI's) are frequently administered as a pre/post assessment. A CI is a multiple-choice exam that emphasizes conceptual understanding over computational skills and has wrong answers designed to match common misconceptions. Inspired by Hake's [1998] pioneering study of pre/post test data from the Force Concept Inventory [Hestenes *et al.*, 1992], we developed the Signals and Systems Concept Inventory (SSCI). In almost two decades since its development, scores of instructors have administered the SSCI to thousands of students in dozens of countries. Analyzing pre/post data from sixty-two signals classes at multiple institutions, we found students in active learning classes learned significantly more than students in lecture classes. Our findings are consistent with both Hake's pioneering study and with Freeman *et al.*'s 2014 meta-analysis of 225 studies that found that active learning classes reduced the failure rate by one-third.

1:25

2pED2. A new pre/post test to assess student mastery of introductory level acoustics and wave mechanics. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu)

The Force Concept Inventory is a widely used multiple-choice test that enables physics educators to assess the ability of students to apply basic concepts associated with Newton's laws and to identify specific misconceptions held by students. When administered at the beginning of a course in introductory physics, as well as at the end, the FCI provides a reliable measure of the *gain* in student understanding. The usefulness of the FCI, and its ease of use, has spurred the development of similar "inventories" and "baseline tests" for other subjects in the physical sciences, such as the Fluid Mechanics Concept Inventory, the Conceptual Survey in Electricity and Magnetism, and the Thermodynamics Concept Inventory. This presentation will describe a new concept inventory test designed to assess

students' conceptual understanding of the behavior of oscillators and mechanical waves, their understanding of the measurement and perception of sound, and their ability to interpret graphical representations of sound. Following the FCI model, this new Sound and Vibration Concept Inventory forces students to compare correct answers to those based on common misconceptions. Preliminary efforts to assess the reliability of this test will be described.

1:45

2pED3. Assessing the concept of resonance through the lens of discipline-based educational research. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

In the last few decades, many science and engineering fields have developed educational research programs specific to their fields. These discipline-based educational research programs are critical for improving the learning environments of the next generation of scientists and engineers. Acoustics is a highly interdisciplinary field, drawing from a wide variety of science and engineering fields. Although no major acoustics educational research program currently exists, acoustics educators should be aware of discipline-based educational research findings which most closely align to their particular branch of acoustics. Educational research can inform educators on how students develop content expertise as well as grow in their thinking about the practice of science and engineering. One of the ways that discipline-based educational research informs educators about the way students think about science content and practices is through the use of assessment instruments. Some assessment tools related to acoustics have been developed by the physics education community, although the selection remains somewhat limited. An attempt to develop an instrument for assessing the concept of resonance as well as the challenges for validating the instrument is presented.

2:05

2pED4. Using and assessing mechanical wave tutorials in introductory physics. Jack Dostal (Phys. Dept., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

The Tutorials in Introductory Physics curriculum produced by the University of Washington Physics Education Group includes tutorials on mechanical waves. The tutorials are written in the style of a Socratic dialogue on topics like superposition, reflection, and transmission of waves. These tutorials have a physics education research basis, leading the student to address common difficulties and misconceptions that have been broadly demonstrated by other students. The tutorials also come with homework assessments and pretest/posttest questions. I will describe my use of these and other tutorials in my own classes, both in standard calculus-based introductory physics courses, as well as in a non-major physics of music class.

2:25

2pED5. Measuring educational outcomes for ABET accreditation of the undergraduate acoustical engineering programs at the University of Hartford. Eoin A. King and Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

Located in West Hartford, CT, USA, the University of Hartford has two ABET-accredited acoustical engineering programs: the Bachelor of Science in Mechanical Engineering (BSME) with Acoustics Concentration and a unique interdisciplinary Bachelor of Science in Engineering with a major in Acoustical Engineering & Music. The national Accreditation Board for Engineering & Technology accreditation process requires that an institution identify and provide evidence of Program Educational Objectives, Student Outcomes, Required Curricular Components, Relevant Facilities, Faculty Expertise, and Methods of Continuous Improvement. The differences and key points of these criteria will be discussed, along the manner by which each are measured. The results of a recent ABET accreditation visit will be described.

Contributed Paper

2:45

2pED6. Imperatives of acoustics and musical instrument acoustics education to the Nigerian educational system in tertiary institutions. Stephen G. Onwubiko, Adebowale O. .Adeogun (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com), Joy O. Nnabuchi (Dept. of Philosophy, Univ. of Abuja Nigeria, Gwagwalada, Abuja, Nigeria), Tobi E. Kemewerigha (College of Education Akamkpa Cross-River, Calabar State, Nigeria, Akakpa CrossRiver, Akakpa, Nigeria), and Ikechukwu E. Onwubiko (Dept. of Civil Eng., Federal Polytechnic Owerri, Owerri, Imo, Nigeria)

This paper discusses about the imperatives of acoustics and musical instrument acoustics education to the Nigerian educational system. It identifies the prospects, the long term educational effects, problems and proffered possible solutions to them. In achieving its objectives the study uses ethnographic and qualitative methods with simple percentages for eliciting and collation of data. The paper suggests that the Acoustics societies work with our curriculum planners, and the government to implant acoustics and its education to tertiary institution in Nigeria. The research questions raised guided the study. It proposes as part of its recommendations that other acoustical bodies provide necessary facilities and personnel for acoustics and its education to thrive as a core subject or vocational subject; and that the larger society must become educated on the usefulness of acoustics as a career subject worth pursuing by scholars.

3:00–3:30 Panel Discussion

Session 2pID

Interdisciplinary and Women in Acoustics: Panel Discussion: Mentoring Across Differences

Dominique A. Bouavichith, Cochair
Linguistics, University of Michigan, 611 Tappan St., #455C, Ann Arbor, MI 48109

Evelyn M. Hoglund, Cochair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43204

Kelly L. Whiteford, Cochair
Psychology, University of Minnesota, 75 East River Parkway, Minneapolis, MN 55455

Chair's Introduction—1:00

Panel Discussion: Mentoring Across Differences. Dominique A. Bouavichith (Linguist, Univ. of Michigan, 611 Tappan St, #455C, Ann Arbor, MI 48109, dbouavichith@gmail.com), Evelyn M. Hoglund (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), and Kelly L. Whiteford (Psych., Univ. of Minnesota, Minneapolis, MN)

A panel of educators in acoustics will discuss issues related to mentoring undergraduate and graduate students, with a focus on practices for creating successful mentoring relationships across differences. The panelists will briefly introduce themselves and describe their experiences as educators and mentors. The panelists will then answer questions from the audience.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 SALON C (VCC), 1:00 P.M. TO 5:00 P.M.

Session 2pNS

Noise, Structural Acoustics and Vibration, and Architectural Acoustics: Noise and Vibration from Fitness Activities

Matthew V. Golden, Cochair
Pliteq, 4211 Yonge St., North York, ON M2P 2A9, Canada

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Fitness centers in mixed-use development: Examples from practice. Benjamin Markham, Ethan Brush, and Robert Connick (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Spin classes utilize sound systems that generate levels in excess of 115 dBA, with subwoofers to match. Gyms located in office buildings feature Olympics style weight lifting technique that involves dropping 200 lb (or more) barbells from shoulder or overhead positions. Medicine balls get thrown at walls and floors. More "typical" sources of airborne and structureborne noise—treadmills, aerobics classes, weight machines, and the like—contribute to the din. The authors will present experiences and lessons learned from actual field experience with fitness centers located in commercial, residential, and mixed-use developments. Data will be presented on the effectiveness of mitigation methods installed in real buildings to attenuate airborne and structureborne noise. The session will include a

discussion of the various types of fitness franchises and associated acoustic concerns, illustrated by case study examples from the authors' acoustical consulting experience.

1:25

2pNS2. Further testing on the efficacy of inertia base weight training stations for CrossFit weight drops. Angela Waters (Kinetics Noise Control, 6300 Irelan Pl., Pataskala, OH 43017, awaters@kineticsnoise.com) and Scott W. Smith (Ballentine Walker Smith, Inc., Kennesaw, GA)

In an attempt to mitigate the adverse impact and vibration effects of CrossFit weight drops in a wood frame house in South Carolina, a concrete inertia base system was designed, constructed, and installed in 2013. The homeowners have been pleased with the system's performance, reporting major reductions in the felt vibration throughout the house. In 2017 initial testing was conducted that showed significant reduction in vibration in an adjacent corridor. This presentation reports on the results of subsequent and more detailed testing at remote locations in the house. New equipment became available which allowed the vibration levels on the inertia base and at the remote locations to be recorded simultaneously.

1:45

2pNS3. Specialty fitness centre noise issues—Case study. Brigette Martin (none, 1200 Lynn Valley Rd., Vancouver, BC v7j2a2, Canada, martin@bkl.ca)

This presentation will include a collection of case studies focusing on specialty fitness centres. There has been growth in the fitness industry for niche fitness facilities, many of which provide instructed classes in small spaces. These facilities can be located in mixed use buildings, where noise from fitness activities can result in complaints from other users in the building. This presentation provides a summary of some projects relating to these spaces, outline the noise issues, noise investigation, and solutions developed for the spaces. We have been invited to present at the Noise and Vibration from Fitness Activities session.

2:05

2pNS4. Long-term monitoring of fitness activity in a commercial gym. Samantha Rawlings, David W. Dong, and John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com)

Noise and vibration from activity in fitness facilities is a common source of complaints in residential, commercial, and mixed use buildings. Most studies have focused on developing and measuring mitigation of the vibration from the activity. However, there is limited information on the characteristics of fitness activity and its effect on acceptability. It is assumed that the complaints will increase with the frequency of occurrence of the events, but there is little information on the temporal distribution of typical fitness activity. There is therefore a need for long term measurements of a large number of fitness facilities, comparable to noise data that have been collected around sources such as roadways and airports. Here, we begin to address this need and report results of long-term vibration monitoring of a commercial fitness facility.

2:25

2pNS5. Determining a relationship between mockup size and transmitted vibration. Paul Gartenburg (Eng., Pliteq Inc, 4211 Yonge St., Toronto, ON M2P2A9, Canada, pgartenburg@pliteq.com) and Matthew V. Golden (Eng., Pliteq Inc., Washington, District of Columbia)

Testing fitness flooring mockups is sometimes required to prove the efficacy of a fully installed floor. Constructing these *in-situ* mockups can be labour intensive and costly. Keeping mockup sizes low is beneficial from these standpoints but can be detrimental from an acoustical standpoint. The purpose of this paper is to understand the relationship between mockup size and transmitted vibration as a result of heavy weight impacts. To achieve this, mockups of varied sizes and buildups ranging from $2' \times 2'$ to $10' \times 0'$ were tested and resultant vibrations in the structural slab were recorded by two accelerometers adhered to the slab. This paper analyzes the recorded results for both locally reactive and resonantly reactive floors to provide insight in determining what mockup size is appropriate without significantly compromising the validity of the results.

2:45–3:00 Break

3:00

2pNS6. Investigation of vibration reduction methods from fitness activity. David W. Dong, John LoVerde, Richard Silva, and Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Noise and vibration from activity in fitness facilities, in particular drops of heavy weights, is a common source of disturbance and complaint in residential, commercial, and mixed-use building types. There is no standardized method for evaluating the reduction in noise or vibration provided by these products and systems. 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 (ΔL_V) achieved due to the insertion of the products. Preliminary results indicated that the ΔL_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 with measurements with additional materials and sources, and preliminary analytical investigations of the effects of mitigation methods.

3:20

2pNS7. Further prediction of heavy weight drops on resilient sports floors in existing buildings. Matthew V. Golden (Pliteq, 4211 Yonge St., North York, ON M2P 2A9, Canada, mgolden@pliteq.com) and Paul Gartenburg (Pliteq, Toronto, ON, Canada)

Currently, the assessment of noise and vibration due to heavy weight drops in an existing building can be cumbersome and time consuming, have low repeatability, and does not allow for evaluation of additional solutions not present at the time of testing. This paper will present further development of a method to solve those problems and is broken up into three parts. The first part is the laboratory assessment of the force pulse that a given weight will exert onto a resilient floor. The second part is a method to quickly obtain the in-situ transfer function between the force injected into the building and the resulting vibration elsewhere in the building. The third part combines the previous two items to predict the acceleration in a building due to any arbitrary combination of impact source and resilient floor. The results obtained will be compared to on-site weight drop measurements in order to qualify the predictive model. Extensions of the model into other structures, types of outputs (vibration vs sound pressure levels), and octave band vs narrow band results will be discussed.

3:40

2pNS8. Assessment and remediation of treadmill generated vibration increased due to a building modification. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Structural modifications to an existing multi-story building installed to increase the load bearing capacity of a floor resulted in a noticeable increase in floor vibration from treadmills used in a fitness room on the floor above. Results from vibration measurements were combined with a finite element analysis (FEA) model of the building dynamics to determine the dynamic force generated by the treadmills in use. The FEA model and the calculated dynamic force were used to evaluate the effectiveness of options for vibration isolating the treadmills to reduce vibration to acceptable levels in the occupied office below.

Contributed Papers

4:00

2pNS9. Effects of noise exposure on hearing health evaluated through short interval otoacoustic emission monitoring: Preliminary results with low and moderate noise exposure groups. Vincent Nadon (École de technologie supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, vincent.nadon@etsmtl.ca) and Jeremie Voix (École de technologie supérieure, Montréal, QC, Canada)

Industrial workers are often exposed on a daily basis to noise doses that put them at risk of Noise-Induced Hearing Loss (NIHL), which remains the largest cause of indemnity in North America industries. To improve occupational hearing loss prevention programs, a method to continuously measure hearing fatigue using otoacoustic emissions (OAE) has already been proposed by the authors using a portable and robust OAE system designed for noisy industrial use. The effectiveness in the field of the aforementioned method is examined by comparing a control group of unexposed subjects in laboratory and noise exposed subjects in the field. All participants are equipped with an hearing protection earpiece which includes the usual OAE measurement components as well as a microphone mounted on the outer faceplate of the earpiece. At regular intervals over the course of the day, the growth function of distortion product OAEs is measured for both groups while simultaneously measuring noise levels inside and outside the ear. Medial olivocochlear reflexes are also measured pre and post exposure to monitor other potential effects on hearing. Preliminary results of this study are presented and analyzed in the broader context of the dose-response relationship that could indicate individuals' risk to develop NIHL.

4:15

2pNS10. Short-term annoyance due to construction work noise in a laboratory context. Jae Kwan Lee, Seo I. Chang (Energy Environment System Eng., Univ. of Seoul, 309, 2nd Eng. Bldg., 163, Seoulsiripdae-ro, Dongdaemun-gu, Seoul 02504, South Korea, jklee645@gmail.com), Jae Woong Jang, and Soo Il Lee (Mech. and Information Eng., Univ. of Seoul, Seoul, South Korea)

Among the various noises that cause environmental disputes, the noise of the construction site is particularly problematic. Psycho-acoustic experiment was used to develop an "annoyance model" due to construction site noise. In the experiment, four noises recorded at the construction site were used. There are two continuous noises and two impact noises. Some people were recruited for experiment and the classification was performed primarily. The subjects were classified into various criteria such as gender, age, sensitivity, disease, and the results were compared. The noise converted to 35–80dB (A) was randomly played. During the experiment, the annoyance was judged by the 11 point scale for every noise. The physical analysis of the sound used in the experiment was performed for the development of the annoyance model besides the auditory experiment. Annoyance model was developed by multiple regression analysis using Leq, Lmax, sound quality indices (loudness, sharpness, roughness, and tonality), and questionnaire score. Also, it was confirmed whether there was any difference in the annoyance model for each subject group.

4:30

2pNS11. The sound insulation performance of mass timber building elements. Jeffrey Mahn (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1C4N4, Canada, jeffrey.mahn@nrc-cnrc.gc.ca)

Mass timber constructions have been gaining popularity in Canada since provincial building codes were changed to allow for taller buildings of mass timber elements. Common mass timber elements include cross-laminated timber, nail-laminated timber, dowel-laminated timber, and glued laminated timber. Many of these elements have been in use for decades, but much still needs to be learned in terms of the transmission of structure-borne noise between elements and the transmission loss and radiation efficiency of the elements with and without linings. Recent and ongoing studies on cross-laminated and nail-laminated elements conducted at the National Research Council have investigated many of the unknowns and a summary of the results are presented in this paper.

4:45

2pNS12. Characterization, analysis, and noise control measures of a mechanical room. Deepak Deokant Jha, Joonhee Lee, and Mohammed Zaheeruddin (Bldg., Civil & Environ. Eng., Concordia Univ., E.V. 16.255, 1455 De Maisonneuve Blvd. W., Montreal, QC H3G 1M8, Canada, deepak.jha0509@gmail.com)

The acoustic environment of a mechanical room in buildings has received little attention because the space is occasionally occupied by building maintenance staff. However, the room typically includes the loudest noise sources in buildings. It is necessary to investigate noise exposure in the mechanical room for the staff. The aim of this study is, thus, to assess noise levels in the room by examining acoustic characteristics of the noise sources and mapping noise levels produced by the sources. This study also aims to provide optimal solutions to decrease the noise levels being produced. The framework consists of (1) identifying the major locations at which noise is being produced; (2) measurement of noise levels; and (3) performing noise level analysis of each activity. These findings will provide useful information to establish effective noise management plans for mechanical rooms in buildings.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 RATTENBURY A/B (FE), 1:00 P.M. TO 4:20 P.M.

Session 2pPA

Physical Acoustics, Architectural Acoustics, Noise, Signal Processing in Acoustics, and Underwater Acoustics: Challenges in Computational Acoustics

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, 72 Lyme Rd., Hanover, NH 03755-1290

Matthew G. Blevins, Cochair
U.S. Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822

Chair's Introduction—1:00

Invited Papers

1:05

2pPA1. Numerical modeling of ultrasound propagation in heterogeneous media using a mixed domain method. Yun Jing and juanjuan Gu (North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

A mixed domain method (MDM) is presented in this paper for modeling one-way linear/nonlinear wave propagation in biological tissue with arbitrary heterogeneities, in which sound speed, density, attenuation coefficients, and nonlinear coefficients are all spatial varying functions. The MDM is based on solving a Westervelt-like equation. We validated the algorithm by studying a number of one-dimensional and two-dimensional problems. Overall, this study demonstrates that the MDM is a computationally efficient and accurate method when used to model wave propagation in biological tissue with relatively weak heterogeneities. We also proposed methods to improve the algorithm for moderately heterogeneous media.

1:25

2pPA2. High frequency propagation in a waveguide with an expanding section. Jerry H. Ginsberg (Dunwoody, GA) and David Feit (ASA, 1305 Walt Whitman Rd, Melville, NY 11747, dfeit@acousticalsociety.org)

A waveguide whose walls have a constant local impedance is readily analyzed if its cross section is uniform with a standard shape. This is not true if the cross section expands or contracts. The (one-dimensional) Webster horn equation typically is used for low frequency propagation in such configurations, but doing so restricts consideration to situations where the size of the cross section changes slowly over the scale of a wavelength. Somewhat paradoxically, a waveguide consisting of a sequence of different sized uniform

segments has been the subject of numerous analyses that are not limited to low frequencies. The present work extends one such analysis, which uses a collocation technique at discontinuous junctions, to a two-dimensional configuration in which a transition section expands linearly. The field in each section is described as a modal series, whose terms are the product of a transverse standing mode and two phasors for axial propagation. Cartesian coordinates are used to describe the field within the uniform sections, whereas a polar coordinate description of the field in the expansion section facilitates satisfying rigid wall conditions. Various schemes for arranging the collocation points are discussed, and convergence of the series is examined.

1:45

2pPA3. Improving the speed of acoustic propagation modeling, for sonar training applications, with a 3D Gaussian ray bundling model. Sean M. Reilly (Physical Sci. and Systems (PS2), Raytheon BBN Technologies, 127 John Clark Rd., Middletown, RI 02842, sean.m.reilly@raytheon.com) and Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

This research improves the calculation speed of acoustic propagation modeling, for sonar training applications in littoral environments, by implementing a 3D Gaussian ray bundling model in the geodetic coordinates of the environmental databases. Tactical decision aids currently transform the three-dimensional environment into two dimensional radials, then compute acoustic propagation as a function of range and depth. This allows them to analyze millions of range, depth, and bearings for potential targets locations in a few minutes. But, it assumes that the cost of transformation is small compared to the speed benefit of computing propagation in Cartesian coordinates. This assumption is violated in applications where the geometry of the sensors and targets is rapidly evolving, results are strongly dependent on location, and answers must be produced faster than real time. Instead of relying on advancements in computer hardware, this research focuses on the development of a new algorithm to compute bistatic, broadband, transmission loss, and reverberation for 100 active sonar acoustic contacts on commodity laptop hardware, at rates ten times faster than the speed of sound, with no measurable impact on accuracy. [Work supported by the High Fidelity Active Sonar Training (HiFAST) Project at the U.S. Office of Naval Research.]

2:05

2pPA4. Optimizing acoustic diffusion of an architectural feature using finite-difference time-domain. Laura C. Brill and John T. Strong (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, lbrill@thresholdacoustics.com)

Parametric design is gaining in popularity as a design tool; however, rigorous analysis of the acoustic characteristics of parametrically designed surfaces typically requires testing physical mockups of a small sample of candidate designs. A computational method using a combination of finite-difference time-domain (FDTD) analysis and a genetic optimization algorithm has been developed to analyze a large number of candidate designs and select the best performers. In this case study, an architect's aesthetic vision was translated into geometric parameters used by a genetic algorithm to optimize acoustic diffusion performance. FDTD was used to simulate an impulse response and analyze the frequency response of reflective/diffusive geometry with a resolution of up to 11 kHz. The process utilized GPU computing clusters available through Amazon Web Services (AWS) to accelerate the analysis. The current availability and relative cost-effectiveness of cloud computing resources makes wave-based acoustic analysis a more viable and efficient option. The methods and resources discussed in this paper allow acoustic performance to be integrated into the architectural optimization process thereby making acoustic performance an active consideration in an iterative design approach.

2:25

2pPA5. Optimized geospatial tool for ambient soundscapes. Michael M. James, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St, Ste. 700, Asheville, NC 28801, michael.james@blueridgeresearch.com), Mark K. Transtrum, Kent L. Gee, Katrina Pedersen, and Brooks A. Butler (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Ambient soundscape data are a critical component of numerous applications such as evaluating the environmental noise impacts of proposed activities and studying the physical and psychological impacts of a person's acoustic environment. The current approach to characterize ambient soundscapes requires intensive field measurement programs. A soundscape prediction tool capable of modeling the soundscape over large spatial regions and timeframes is needed to provide an alternative to intensive field measurements. To address this need, a machine learning based soundscape model trained with acoustic data and geospatial layers has been developed. The model's capability to generate A and flat-weighted exceedance levels across space, time (hourly, daytime, and nighttime), and frequency (one-third octave bands) will be demonstrated. [Work funded by an Army SBIR.]

2:45

2pPA6. Machine learning-based prediction of outdoor ambient sound levels: Ensemble averaging and feature reduction. Katrina Pedersen, Kent L. Gee, Mark K. Transtrum, Brooks A. Butler (Brigham Young Univ., N283 ESC, Provo, UT 84602, katrina.pedersen@gmail.com), Michael M. James, and Alexandria R. Salton (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Outdoor ambient sound levels can be predicted from machine learning-based models derived from geospatial and acoustic training data. To improve modeling robustness, median predicted sound levels have been calculated using tuned models from different supervised machine learning modeling classes. The ensemble-based model reduces errors at training sites for both overall levels and spectra, and produces more physically reasonable predictions elsewhere. Furthermore, the spread in the ensemble provides an estimate of the modeling accuracy. An initial analysis of feature importance metrics suggests that the number of geospatial inputs can be reduced from 120 to 15 without significant degradation of the model's predictive error, as measured by leave-one-out cross validation. However, the predictions from the reduced-feature modeling may be less physical in certain regions when all differentiating geospatial features are removed. These results suggest the need for more sophisticated data collection and validation methods.

3:05–3:20 Break

3:20

2pPA7. Optimizing calculation points using Gaussian process regression. Matthew Kamrath (CSTB, 72 Lyme Rd., Hanover, New Hampshire 03755, Matthew.J.Kamrath@erdc.dren.mil), Philippe Jean, Julien Maillard (CSTB, Saint-Martin-d'Hères, France), and Judicaël Picaut (IFSTTAR, Bouguenais Cedex, France)

Many applications in computational acoustics calculate a value (e.g., sound pressure level) at many different locations to approximate the value throughout a region using interpolation. Often, the points are uniformly or exponentially spaced without a rigorous procedure to optimize the locations because directly minimizing the interpolation error is too computationally expensive. Instead, the interpolation error can be indirectly reduced by minimizing the maximum variance estimated using Gaussian process regression. This approach is less expensive because the variance at a point is not a function of the value at that point. Thus, each evaluation of the objective function (i.e., the maximum variance) does not require additional acoustical computations, which mitigates the cost of the objective function. As an example, this procedure is applied to an outdoor sound propagation case to approximate the insertion loss of a 3 m tall T-shaped barrier compared to 3 m tall straight barrier, which are modeled using the boundary element method. In this case, the optimized locations have a smaller interpolation error than uniformly and exponentially distributed points.

3:35

2pPA8. A highly efficient approach to model acoustics with visco-thermal boundary losses. Martin Berggren, Daniel Noreland, and Anders Bernland (Dept. of Computing Sci., Umeå Univ., Campustorget 5, Umeå 90187, Sweden, martin.berggren@cs.umu.se)

For devices such as hearing aids, microphones, micro loudspeakers, and compression drivers, thermal and viscous boundary layer effects are often highly noticeable. These effects can be modeled in the linear regime by the linearized, compressible Navier-Stokes equations. However, the need for resolution of the very thin boundary layers typically makes numerical solutions of these equations computationally very expensive. Based on a boundary-layer analysis, we have derived for the pressure Helmholtz equation what appears to be a new boundary condition that accurately takes visco-thermal boundary losses into account. The model is valid when the wavelength and the minimum radius of curvature of the wall is much larger than the boundary layer thicknesses. In the special case of sound propagation in a cylindrical duct, the model collapses to the classical Kirchhoff solution. We assess the model in the case of sound propagation through a compression driver, a kind of transducer that is commonly used to feed horn loudspeakers. The transmitted power spectrum through the device calculated numerically using our model agrees extremely well with computations using a hybrid model, where the full linearized, compressible Navier-Stokes equations are solved in the narrow regions of the device and the pressure Helmholtz equations elsewhere. However, our model needs two orders of magnitude less memory and computational time than the more complete model.

3:50

2pPA9. Detection of bubble movements via wave phase conjugation. Amir Modarreszadeh, Evgeny Timofeev (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A0C3, Canada, amir.modarreszadeh@mail.mcgill.ca), Alain Merlen, Olivier Bou Matar, and Philippe Pernod (Electronics, Microelectronics and NanoTechnol., Univ. of Lille 1, Villeneuve d'Ascq, France)

Wave Phase Conjugation (WPC) means reversing the propagation of waves so that initial spatial distributions of amplitudes and phases are conserved. One of the methods to conjugate ultrasonic waves is to modulate the speed of sound in a solid conjugator by an alternating electromagnetic field. The detection of bubble motions in a fluid to be considered in this work is among potentially interesting and omnipresent WPC applications in industry. As the ultrasonic waves have short wavelengths, high-order in space and time numerical methods are required for modeling. In this work, a modified version of the Nodal Discontinuous Galerkin method, which is based on the non-collocated solution and flux bases, is implemented for wave propagation in solids and liquids (for both linear and non-linear flow regimes) in an axisymmetric geometry. Being assured of the accuracy and performance of the numerical technique by evaluating some representative test cases, the detection of bubble motion and growth in a flow field using WPC is simulated with the inclusion of all elements of the WPC process: the transducer, the conjugator, and the bubbly liquid itself. The developed methodology and results can be used to design and improve measurement devices based on the WPC phenomenon.

4:05

2pPA10. Target-oriented waveform inversion. Tianci Cui, James E. Rickett (Geophys., Schlumberger Cambridge Res., High Cross, Madingley Rd., Cambridge CB30EL, United Kingdom, tcui2@slb.com), and Ivan Vasconcelos (Earth Sci., Utrecht Univ., Utrecht, Netherlands)

Full-waveform inversion (FWI) has demonstrated increasing success in estimating high-resolution medium parameters, which are obtained by minimizing the difference between synthetic and real data on the acquisition surface. In the standard approach, the whole propagation domain needs to be resolved. This highly nonlinear problem is usually solved via a gradient-based local optimization approach, which can be computationally very expensive. We propose target-oriented FWI utilizing novel redatuming techniques. From surface measurements, Marchenko redatuming retrieves full-waveform up- and downgoing wavefields inside the medium at no significant computational cost. We choose two datum locations inside the medium, one above and one below the target volume. Marchenko-redatumed wavefields at both target-enclosing boundaries consist of in- and outgoing wavefields. The outgoing wavefields are the response of the ingoing wavefields being scattered from the target volume only, without interference from the overburden or underburden. We conduct FWI by finding a target model whose response to the ingoing wavefields fit the outgoing wavefields best. The computational cost of targeted FWI, which involves solving the wave equation in the target volume only, will lead to savings over the traditional full-volume approach. We validate our method numerically on a 2D acoustic model.

Session 2pPP

Psychological and Physiological Acoustics and Speech Communication: Speech Perception in Children with Hearing Impairment

Kelly N. Jahn, Cochair
Speech and Hearing Sciences, University of Washington, 1417 NE 42nd Street, Box 354875, Seattle, WA 98105-6246

Mishaela DiNino, Cochair
Speech and Hearing Sciences, University of Washington, 1417 NE 42nd St., Seattle, WA 98105

Chair's Introduction—1:00

Invited Papers

1:05

2pPP1. Optimizing speech recognition in realistic environments for children who wear hearing aids. Ryan W. McCreery (Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ryan.mccreery@boystown.org), Elizabeth Walker (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), and Marc Brennan (Special Education and Commun. Disord., Univ. of Nebraska - Lincoln, Lincoln, NE)

Children with hearing loss are immersed in listening situations with poor signal-to-noise ratios and high levels of reverberation. Children's ability to effectively communicate in acoustically challenging listening situations has an impact on language development and academic and social functioning. Children with hearing loss who use hearing aids perform more poorly than peers with normal hearing when listening in noise and reverberation. However, speech understanding among children with hearing loss is heterogeneous and most previous research has focused primarily on the effect of a child's degree of hearing loss derived from the audiogram. Our research explores additional factors that contribute to individual differences in speech recognition in adverse acoustic conditions for children with hearing loss. This work has focused on three factors that explain individual differences in speech recognition in noise among children who use hearing aids: 1) the degree to which speech audibility is restored by the hearing aid(s), 2) the child's language abilities, and 3) the child's cognitive abilities, including working memory and selective attention. The relationships between speech recognition and these factors, as well as the clinical implications of these findings, will be described.

1:35

2pPP2. A functionally relevant measure of spatial release from masking in children with bilateral cochlear implants. Z. Ellen Peng and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, z.ellen.peng@wisc.edu)

The ability to segregate speech from noise is an important skill for children as they navigate everyday noisy environments. Spatial release from masking (SRM) is a measure of benefit observed when target speech is co-located versus spatially separated from maskers. SRM is typically studied with target-maskers in front versus maskers displaced to fixed angles towards the side. Children fitted with bilateral cochlear implants (BiCIs) typically show smaller SRM than children with normal hearing (NH), even for target-masker angular separation of 90°, and when monaural head shadow cues exist. Here, we used an open-set corpus that has high informational masking and better mimics realistic listening situations for children. First, SRM was measured by adapting target-masker separation, and quantified as the minimal angular separation needed to achieve 20% improvement in target intelligibility. Second, SRM was measured for target-masker separation of 180°. Preliminary results suggested that, in the BiCI group, all children achieved 20% SRM with a target-masker angular separation smaller than 180°. With the full 180° target-masker separation, children in the BiCI group achieved a much larger SRM (6–8 dB) than previously reported while NH children's SRM was larger. Acoustic cues available through CI processors will be discussed. [Work funded by NIH-NIDCD.]

Contributed Paper

2:05

2pPP3. Vowel confusion patterns in children and adults with cochlear implants. Kelly N. Jahn, Mishaela DiNino, Matthew Winn, Jacob Hulswit, and Julie G. Arenberg (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Box 354875, Seattle, WA 98105-6246, jahnk@uw.edu)

Suboptimal interfaces between cochlear implant (CI) electrodes and their target neurons may distort the spectral information that is important for vowel identification. The goal of this investigation was to relate vowel confusion patterns of CI users to their site-specific (i.e., frequency-specific) variation in electrode-neuron interface across the electrode array. The electrode-neuron interface is estimated using single-channel detection thresholds with a spatially focused electrode configuration; elevated thresholds indicate channels with relatively poorer frequency transmission. We hypothesized that higher focused thresholds on channels transmitting low frequency (<983 Hz) information would be associated with errors in vowel height, whereas higher thresholds on channels transmitting mid- and high-frequency (>984 Hz) information would be related to vowel advancement errors. Vowel recognition was assessed with a closed set of medial vowels in 4-talker babble noise. Focused thresholds were measured on electrodes 2–15. Vowel confusions were examined according to the direction of errors in 2-dimensional physical articulatory and acoustic F1-F2 space. Results showed that higher average focused thresholds on the 3 most basal electrodes tested (2327–4630 Hz) were associated with perception of front vowels as more back. These findings suggest that suboptimal electrode-neuron interfaces could systematically affect CI listeners' perception of vowel characteristics.

Invited Papers

2:20

2pPP4. Remote microphone system use in the homes of children with hearing loss. Carlos R. Benitez-Barrera, Gina Angley, and Anne Marie Tharpe (Hearing and Speech Sci., Vanderbilt Univ., 2031 Convent Pl, 2031, Nashville, TN 37212, carlos.r.benitez@vanderbilt.edu)

Remote microphone systems (RMS) are known to improve speech recognition skills of children with hearing loss in settings where noise and distance are present (e.g., the classroom, the home; Bertachini, 2016). However, although RMSs are widely recommended in the classroom setting, very few and inconclusive studies have examined the possible language benefits related to the use of RMS use in the home environment. In this presentation, we will discuss a series of projects in which we investigated the effects of home-use of an RMS on the communication of 10 families with young children with hearing loss. Language Environmental Analysis (LENA™) recorders were used during two consecutive weekends (one weekend with the remote microphone and one without). Caregiver talk, child-directed speech, and other caregiver communication strategies were compared across both weekends. Caregiver perceptions related to RMS use will also be discussed.

2:50–3:10 Break

3:10

2pPP5. Spectral modulation detection in adolescents with normal hearing or cochlear implants predicts some language skills, but not others. Susan Nittrouer, Joanna H. Lowenstein, and Donal Sinex (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2147, Gainesville, FL 32610, snittrouer@phhp.ufl.edu)

Previous outcomes of an ongoing longitudinal study involving children with cochlear implants (CIs) have revealed that these children demonstrate especially large deficits in two areas: (1) language skills dependent upon phonological structure; and (2) speech-in-noise recognition. Morphosyntactic skills appear closer to typical. We have hypothesized that both the phonological deficit and the speech-in-noise problems arise from poor spectral resolution. To test this hypothesis we measured children's spectral modulation detection (SMD). Participants were14-year-old children with normal hearing (49) or with CIs (42). A 3-interval, forced-choice procedure was used in which two stimuli were unmodulated noises and one was a noise modulated at a rate of 0.5 ripples per octave (rpo). The 70.7% threshold for SMD was estimated adaptively. Other measures included tests of phonological awareness and processing, vocabulary and syntactic abilities, and speech recognition in single- or multi-talker babble, as well as in speech-shaped noise. Children with CIs had slightly higher SMD thresholds, and poorer performance on all other measures. For both groups, SMD thresholds were related to phonological skills and speech-in-noise recognition, but not to vocabulary or syntactic abilities. Outcomes were interpreted as suggesting that phonological sensitivity and speech-in-noise recognition depend upon spectral resolution, but morphosyntactic skills do not.

Contributed Paper

3:40

2pPP6. Perception of Mandarin Chinese vowels in young hearing-impaired and normal-hearing children. Changxin Zhang (Dept. of Education and Rehabilitation, East China Normal Univ., Shanghai, Shanghai, China) and Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu)

The study investigated the perception of Mandarin Chinese vowels of 45 congenital hearing-impaired and 45 normal-hearing children aged 4 to 6. There were 30 children at each age group, half of which were hearing-impaired children. All of the hearing-impaired children received hearing aids or cochlear implants before the age of 5. In a picture identification task, listeners identified the target consonant-vowel word among 2–4 contrastive words. The target words and the other contrastive words differed only in consonants. Each target word represented a concrete object and was spoken by a young female native-Chinese talker. Sixteen of the target words ended with monophthong, twenty-two with diphthong and nine with triphthong. Age showed a significant effect on vowel perception for both groups. Normal-hearing children showed significantly better identification of all three types of vowels than hearing-impaired children at the age of 6, whereas the two groups had comparable performance at age of 4 and 5. For hearing-impaired children, a rapid development of diphthong perception occurred between 4 and 5 years old, while a rapid development of monophthong perception between 5 and 6. The effect of hearing loss severity and the use of hearing aids or cochlear implants will be discussed.

Invited Paper

3:55

2pPP7. Your ears never sleep: Auditory processing of nonwords during sleep in children. Adrienne Roman, Carlos Benitez, Sasha Key, and Anne Marie Tharpe (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 904 Erin Ln., Nashville, TN 37221, adrienne.s.roman@vanderbilt.edu)

The auditory system in humans is the only sensory system that remains active for the continuous scanning and processing of information during sleep; however, little research has investigated this phenomenon. This study examined whether a brief exposure to auditory stimuli during sleep could result in memory traces indicated by event-related potentials (ERPs) in children. Twelve preschool children with normal hearing (2–5 years) were presented with three randomly selected nonwords (250 trials; 45–50 dB SPL) for 10 min during a regular nap (sleep state verified by EEG). After the nap, children's memory for the stimuli was evaluated using a passive listening version of the "old/new" ERP paradigm that included three nonwords from the nap exposure (repeated condition, 45 trials), 3 new nonwords (sham repeated condition, 45 trials), and 45 other distinct nonwords (novel condition). Memory for the "nap exposure" words was reflected by increased positive amplitudes at midline parietal locations between 350 and 600 ms compared to the sham and novel conditions. This evidence of the memory trace for the auditory stimuli experienced during sleep can have implications for children with hearing loss who routinely remove their assistive devices during sleep. Preliminary results from children with cochlear implants will be discussed.

TUESDAY AFTERNOON, 6 NOVEMBER 2018 UPPER PAVILION (VCC), 1:00 P.M. TO 2:30 P.M.

Session 2pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics and Vibration (Poster Session)

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

All posters will be on display from and all author will be at their posters from 1:00 p.m. to 2:30 p.m.

Contributed Papers

2pSA1. Remote focusing of elastic waves in complex structures using time reversal of acoustic waves. Maxime Farin, Claire Prada, and Julien de Rosny (Acoust., Institut Langevin, Institut Langevin, 1, rue, Jussieu, Paris, France 75005, France, maxime.farin@espci.fr)

Remotely assessing the state of damage of thin plates or beams within a complex structure is a burning issue in many industrial problems. A time-reversal technique is used to focus a short and localized elastic impulsion on a thin plate using a network of loudspeakers. The generated plate vibration is measured using a laser vibrometer. We built an experimental setup to mimic a complex structure, with several thin plates attached close to each others inside a cylinder. We show that the acoustic method allows us to put a targeted plate in vibration inside the cylinder, without exciting the other plates in the same structure. The time-reversal technique takes advantage of the strong wave reverberation caused by the presence of the complex structure around the plate, which creates virtual sound sources, and improves the focusing on the plate compared to that obtained when the plate has nothing around it. The ratio of plate vibration energy over emitted acoustic energy is about 1%. Finally, we create a small defect on a plate and assess whether the defaults can be detected from changes in the plate vibration modes. Our experimental results are compared with finite elements simulations.

2pSA2. Road pavement defect assessment through vibration analysis inside vehicles. Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl) and Maciej Szczodrak (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

Miniature accelerometer sensors were used for the evaluation of road surface roughness. The device designed for installation in the vehicles is composed of a GPS receiver and of multi-axis accelerometers. Smartphones with built-in accelerometers were also used. Measurement data were collected through the recording of road trips employing 3 car types on diversified surface roughness roads and with varied vehicle speed on each investigated road section. The first step of data processing was the sensor alignment made to achieve proper values of acceleration vector. Subsequently, the influence of the car suspension system to the measurement results was diminished employing a designed filter. The magnitude of coefficients of Gabor transform analysis of accelerometer signals was calculated to discover differences between good and damaged surface. Research results show that road sections quality can be assessed by the applied vibration analysis. Although the precision of low-cost devices may be lower than the application of expensive professional laser profilograph scanning on road pavements, they can help to increase the effectiveness and the coverage of road surface damage detection, through the monitoring of road surfaces with typical cars instead of special test vehicles only. [Research was subsidized by the Polish National Centre for Research and Development and the General Directorate of Public Roads and Motorways within the Grant No. OT4- 4B/AGH-PG-WSTKT.]

2pSA3. Dynamic response sensitivity and variability of a buckling-restrained braced frame under earthquake ground motion. Max D. Magalhaes (Structural Eng., Federal Univ. of Minas Gerais, R. Sao Sebastiao do Paraiso, 305/101 - Itapoa, Belo Horizonte 31710080, Brazil, maxdcm@gmail.com) and Tony Yang (Civil Eng., Univ. of Br. Columbia, Vancouver, BC, Canada)

The main goal of this paper is to examine the variability of deformation, stress and energy absorption of a BRBF (Buckling-Restrained Brace Frame) to some dynamic parameters via a parametric study. This study is aimed at providing not only a better understanding of the vibration transmission mechanism in itself but also to produce a useful set of data which for instance can be used by earthquake engineers as input data for a hybrid analysis. This data might be useful for optimizing vibration insulation in buildings at low frequencies, where the modal behaviour of buildings strongly influences the transmission. The results that are discussed in this paper were obtained via simulations using a modal model. The analysis is based on considering the influence of some variations in the "input" parameters, which are required in the pre-processing stage of a numerical experiment, and on the subsequent vibration transmission mechanisms of typical building configurations.

2pSA4. Eulerian motion magnification applied to structural health monitoring of wind turbines. Sebastian Cygert (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland) and Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

Several types of defects may occur in wind turbines, as physical damage of blades or gearbox malfunction. A wind farm monitoring and damage prediction system is built to observe abnormal vibrations of elements of wind turbine: blades, nacelle, and tower. Contactless methods are developed which do not require turbine stopping. In this work, structural health monitoring of a wind turbine is evaluated using a conversion from the captured and processed video to the acoustic signal, employing the method of Eulerian motion magnification in video. It was assumed that this task can be achieved using a stabilized high-speed video camera only, directed at the wind turbine without any additional sensors mounted on windmill blades or on its body. Moreover, the developed vector sound intensity probe was used for spatial measurements in order to recover the vibration modes of wind turbines. Finally, statistical methods were applied to the processing of computed features reflecting vibrations in order to determine wind turbine technical condition. The developed method was evaluated empirically in real wind farm. [Research was subsidized by the Polish National Centre for Research and Development within the grant "STEO—System for Technical and Economic Optimization of Distributed Renewable Energy Sources," No. POIR.01.02.00-00-0357/16.]

2pSA5. Effects of L2 proficiency on the production of voiceless stops. Ji-Hyun Jeon (Yonsei Univ., Seoul KS013, South Korea, wjswlgus6623@hanmail.net) and Seok-Chae Lee (Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

This study investigates how native speakers and L2 learners of English (L2 learners being native speakers of Korean) produce voiceless stops in English in the following phonological contexts: word-initial vs. -medial, stressed vs. unstressed, and when preceded by /s/. The study also examines the correlation between proficiency of L2 learners of English and the degree of aspiration (represented by VOT) in English voiceless stops. The speech of Korean L2 learners of English rated and categorized into 5 English proficiency levels in Genie Speech Corpus will be used. We will measure VOT of voiceless stops in the aforementioned conditions produced by 5 English speakers and each 5 Korean speakers per level. We expect that highly rated L2 speakers of English will produce longer VOT in stressed syllables than in unstressed syllables, as native speakers of English do. However, as the proficiency level gets lower, L2 speakers will produce it to a lesser degree than native speakers do, presenting a wide variety of VOT length.

2pSA6. Experimental research on fatigue performance of aero-structure under two combined loads. Zhihong Liu (Northwestern PolyTech. Univ., Youyi xilu 127#, Xi'an 710072, China, liuzh@nwpu.edu.cn), Li zhang (Aircraft Strength Res. Inst., Xi'an, China), Xing Liu (Northwestern PolyTech. Univ., Xi'an, China), Dingwen Guo, and Kai Pan (Aircraft Strength Res. Inst., Xi'an, China)

In this paper, a new engineering test method which put forward the initial stage of aircraft design, bearing both acoustic and static loads, is carried out in Lab. From currently test effect, it could accurately simulate the real situation of aircraft panel under hydrostatic load and acoustic load, and the test result shows that the structure presents plastic deformation under the combined action of acoustic load and static load, which differs from the common failure mode of the structure under the action of the single strong noise load. Furthermore, the proposed method could provide guidance to support aircraft type selection.

Session 2pSC

Speech Communication: Speech Perception (Poster Session)

Rajka Smiljanic, Chair
Linguistics, University of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

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

2pSC1. Perception of vowels with missing formant information. Filip Nenadic (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada), Pamela Coulter, Michael Kiefte (School of Commun. Sci. and Disord., Dalhousie Univeristy, 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca), and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

The dominant formant-based model of vowel perception has been challenged by several whole-spectrum approaches. Recent arguments in favor of the importance of cues other than formant frequencies come from a study by Ito *et al*. [J. Acoust. Soc. Am., 110, 1141–1149] in which suppressing either of the first two formants did not radically change the identification of Japanese vowels. The present study replicates the experiment using the larger vowel system of the English language. Visual inspection shows that even when a formant is suppressed, listener responses do not deviate as much as would be expected if formants were the sole cue for vowel identification. However, quantitative analyses indicate that participant agreement in which vowel they heard is significantly lower when they respond to stimuli with suppressed formants. Additionally, the suppressed formant value becomes a less important predictor of vowel identity. These changes in responses become even larger when stimuli (original, F1-suppressed, and F2-suppressed vowels) are presented together rather than in separate blocks. Taken together, these results show that, although formants are not the only correlate to vowel identity, they seem to be the most important.

2pSC2. Short-term, not long-term, average spectra of preceding sentences bias consonant categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Phoneme perception is influenced by spectral properties of surrounding sounds. For example, listeners perceive /g/ (lower F3 onset) more often after sentences filtered to emphasize high-F3 frequencies, and perceive /d/ (higher F3 onset) more often after sentences filtered to emphasize low-F3 frequencies. These biases are known as spectral contrast effects (SCEs). Much of this work examined differences between long-term average spectra (LTAS) of preceding sounds and target phonemes. Stilp and Assgari (2018 ASA) revealed that spectra of the last 500 ms of precursor sentences, not the entire LTAS, predicted biases in consonant categorization. Here, the influences of Early (before the last 500 ms) versus Late (last 500 ms) portions of precursor sentences on subsequent consonant categorization were compared. Sentences emphasized different frequency regions in each temporal window (Early = low-F3 emphasis, Late = high-F3 emphasis, and vice versa) naturally or via filtering. Entire-sentence LTASes predicted that SCEs would not bias consonant categorization, but instead responses were biased by the spectrum of the Late window. This was replicated when the Early window did not emphasize either frequency region but the Late window did. Results endorse closer consideration of patterns of spectral energy over time in preceding sounds, not just their LTAS.

2pSC3. CRSS-LDNN: Long-duration naturalistic noise corpus containing multi-layer noise recordings for robust speech processing. John H. L. Hansen, Harishchandra Dubey, and Abhijeet Sangwan (The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Multi-layer noise refers to scenarios where multiple distinct noise sources are simultaneously active in an audio stream. We collected a corpus named the CRSS long-duration naturalistic noise (CRSS-LDNN) corpus. It contains noise captured from complex daily-life activities using wearable LENA units. The diversity in noise-sources include construction noise, noise like multi-speaker babble, large-crowd noise, vehicle/bus noise on the road, home environment noise, etc. Corpus contains recordings of complex mixtures of these noise types. The babble noise and bus-engine noise present along with occasional impulsive-noise over long-duration is an example of such scenario. The data were collected at 16 kHz sampling rate with 16-bit precision in .wav format. This data would be released to speech community (http://crss.utdallas.edu). During the summer semester, a CRSS student wore a LENA device that was switched ON when multiple noise-sources were present. This corpus was recorded in naturalistic scenarios with uncontrolled mixing of various noise-sources. It provides naturalistic multi-layer noise recordings for evaluation of robust speech algorithms such as speech recognition, speaker diarization and verification, sentiment analysis. It consists of approximately 19 hours noise recordings. The CRSS-LDNN noise is more challenging as compared to existing noise corpora such as NOISEX that contains only single noise.

2pSC4. The role of F2 and F3 in the perception of liquids for Hindi and Mandarin listeners. Phil Howson (Univ. of Oregon, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca), Jessamyn L. Schertz (Univ. of Toronto, Mississauga, Mississauga, ON, Canada), and Irfana M. (Netaji Subhash Chandra Bose Medical College, Jabalpur, India)

Rhotics and laterals are often described as being separated by F3; however, recent work has described lower F2 as a distinguishing quality of rhotics. The current work examines the perception of liquids (rhotics and laterals) by Mandarin and Hindi listeners. Participants completed a forced-choice identification task on a set of 30 stimuli, manipulated from a natural Malayalam production /aɻa/ to vary systematically in F2 and F3. Listeners chose the consonant they heard from a set of approximants (/r/, /l/, /j/, and /w/ or /ʋ/). The results revealed that liquid identification was mostly confined to the mid F2 range, from 1228 Hz to 1728 Hz. Within that range, stimuli with the lowest F2 and F3 values were predominantly identified as rhotics, while those with the highest F2 and F3 values were identified as laterals. The F3 range of 1885 Hz–2135 Hz was ambiguous between rhotics and laterals. The data suggest that F2, as well as F3, plays a key role in rhotic identification, and suggest overlap in the F3 perceptual space for rhotics and

laterals for Mandarin and Hindi listeners. The data suggest a complex relationship between F2 and F3 and liquid perception.

2pSC5. Aging effects on categorical perception of Mandarin tones in noise. Can Xu (Commun. Sci. and Disord., The Univ. of Texas at Austin, School of Foreign Lang., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai, Shanghai 200240, China, 8023xc@sjtu.edu.cn), Yuxia Wang, Xiaohu Yang (School of Foreign Lang., Shanghai Jiao Tong Univ., Shanghai, China), and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

The current study aimed at investigating the aging effect on the categorical perception of Mandarin lexical tones with varied fundamental frequency (F0) contours in noise. Mandarin tone identification and tone discrimination in quiet and noise were measured for younger and older listeners in a categorical perception paradigm where the stimuli continua varied their F0 contour systematically from the level to the rising/falling tones. Results reported that older listeners, in contrast with their younger counterparts, performed with less stimulus-tuned changes in both identification and discrimination functions and smaller peakedness in the discrimination function for both level-rising and level-falling tones. The aging effects on Mandarin tone categoricality were observed in both quiet and noise. Moreover, noise aggravated the aging effects, especially with the high SNR condition. Plus, older listeners' identification and discrimination functions in CP paradigm positively correlated with their performance in general speech identification in noise; such correlation was not found with younger listeners. Our study suggested that older listeners had less categoricality in both identification and discrimination functions for the level-rising and level-falling tones, probably due to the aging-related decline in temporal processing.

2pSC6. Effects of receptive language ability on the neural representation of phonetic category structure. Julia R. Drouin and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs, CT 06269, julia.drouin@uconn.edu)

Speech sound categories have a graded internal structure that reflects typicality of the speech input. Neuroimaging findings reveal dissociable regions for resolving category membership and processing phonetic category structure; frontal regions show increased activation for ambiguous compared to unambiguous exemplars, and temporoparietal regions show increased activation for poor compared to prototypical exemplars, even when category membership is unambiguous. Individuals with developmental language disorder (DLD) show behavioral deficits in some categorization tasks, which may reflect difficulty in organizing auditory information into structured categories. An fMRI paradigm is used to examine whether the neural representation of phonetic category structure reflects receptive language ability. Participants were assigned to either the DLD or control group based on performance for standardized assessments of receptive and expressive language. To assess neural representation of phonetic category structure, participants completed a phonetic categorization task for tokens of *bowl* and *pole* using an event-related, sparse-sampling design; voice-onset-times (VOTs) of the *pole* tokens were manipulated to have values representing an ambiguous exemplar, a prototypical exemplar, and an extreme (i.e., long VOT) exemplar. The analyses will examine whether individuals with DLD show attenuated neural representation of phonetic category structure, and, if so, whether attenuation reflects individual differences in receptive language ability.

2pSC7. Structured phonetic variation facilitates talker identification. Divya Ganugapati and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Individual talkers systematically differ in their phonetic implementation of speech sounds, and listeners use this structure to facilitate language comprehension. Here we test the hypothesis that listeners' sensitivity to talker-specific phonetic variation also facilitates voice processing. Listeners completed training and test phases. During training, listeners learned to associate talkers' voices with cartoon avatars (with feedback). Training stimuli consisted of single-word utterances from two minimal pairs (*gain*, *cane*, *goal*, and *coal*). In one condition, voice-onset-times (VOTs) of the voiceless-initial words were structured such that each talker had a characteristic VOT. In the other condition, listeners heard the same VOTs, but no talker-specific phonetic structure was present. At test, listeners heard all VOT variants for trained and novel words, and they performed the same voice-avatar identification task (without feedback). The results showed that given limited exposure (one training block), learning of the talkers' voices was equivalent between the two conditions. However, given extended exposure (three training blocks), talker identification was improved with respect to both accuracy and processing time for listeners who received structured phonetic variation compared to those who did not. These findings suggest that given sufficient time to learn talkers' idiolects, listeners use structured phonetic variation to facilitate voice processing.

2pSC8. He said, she said: Talker-specific influences on memory for spoken language. Elizabeth O'Brien and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, elizabeth.obrien@uconn.edu)

Memory for spoken language is not a veridical representation of experience. Instead, memory reflects integration across our interlocutors' messages, resulting in robust memory for meaning with relatively poor memory for specific form. Here we test the hypothesis that talker identity influences how spoken language is integrated in memory. Listeners completed encoding and recognition phases. During encoding, listeners heard sentences that contained constituents of four-part semantic units (i.e., "idea sets"). During recognition, listeners heard novel sentences that contained constituents of each idea set in addition to full idea sets and four-part sentences formed by combining constituents across idea sets (i.e., "noncases"). Across experiments, talker was manipulated in terms of (1) the match between encoding and recognition and (2) the number of talkers during encoding. The results to date replicate evidence of integration during encoding; when talker is held constant, listeners show more false memories at recognition for idea sets compared to constituents, and no false memories for noncases. However, false memories decrease when the talker differs between encoding and recognition, and when listeners hear multiple talkers compared to a single talker during encoding. These findings suggest that talker identity provides a critical structure for the integration of spoken language in memory.

2pSC9. The stability of visual–aerotactile effects across multiple presentations of a single token. Sharon Kwan, Megan Keough, Ryan C. Taylor, Terrina Chan, Murray Schellenberg, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, sksharonkwan@gmail.com)

Previous research has shown that the sensation of airflow causes bilabial stop closures to be perceived as aspirated even when paired with silent articulations rather than an acoustic signal [Bicevskis *et al.* 2016, JASA 140(5): 3531–3539]. However, some evidence suggests that perceivers integrate this cue differently if the silent articulations come from an animated face [Keough *et al.* 2017, Canadian Acoustics 45(3):176–177] rather than a human one. Participants shifted from a strong initial /ba/ bias to a strong /pa/ bias by the second half of the experiment, suggesting the participants learned to associate the video with the aspirated articulation through experience with the airflow. One explanation for the above findings is methodological: participants saw a single video clip while previous work exposed participants to multiple videos. The current study reports two experiments using a single clip with a human face (originally from Bicevskis *et al.* 2016). We found no evidence of a bias shift, indicating that the findings reported by Keough *et al.* are not attributable to the use of a single video. Instead, our findings suggest that aero-tactile cues shift consonant perception regardless of the number of recordings presented as long as the speaking face is human.

2pSC10. Perceiving prosodic prominence via unnatural visual information in avatar communication. Ryan C. Taylor, Dimitri Prica, Megan Keough (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, taylryan@gmail.com), and Bryan Gick (Haskins Labs., Vancouver, Br. Columbia, Canada)

Listeners integrate information from simulated faces in multimodal perception [Cohen, & Massaro 1990, Behav. Res. Meth. Instr. Comp. 22(2), 260–263], but not always in the same way as real faces [Keough *et al.* 2017, Can. Acoust. 45(3):176–177]. This is increasingly relevant with the dramatic increase in avatar communication in virtual spaces [https://www.bloomberg.com/professional/blog/computings-next-big-thing-virtual-world-may-reality-2020/]. Prosody is especially relevant, because compared to segmental speech sounds, the visual factors indicating prosodic prominence (e.g., eyebrow raises and hand gestures) frequently bear no biomechanical relation to the production of acoustic features of prominence, but are nonetheless highly reliable [Krahmer & Swerts 2007, JML 57(3): 396–414], and avatar virtual communication systems may convey prosodic information through unnatural means, e.g., by expressing amplitude via oral aperture (louder sound = larger opening); the present study examines whether this unnatural but reliable indicator of speech amplitude is integrated in prominence perception. We report an experiment describing whether and how perceivers take into account this reliable but unnatural visual information in the detection of prosodic prominence. Preliminary evidence suggests that oral aperture increases prominence with differences by sentence position.

2pSC11. Perceiving audiovisual speech articulation in virtual reality. Megan Keough, Ryan C. Taylor, Dimitri Prica, Esther Y. Wong, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mkeough@alumni.ubc.ca)

Listeners incorporate visual speech information produced by computer-simulated faces when the articulations are precise and pre-programmed [e.g., Cohen, & Massaro 1990, Behav. Res. Meth. Instr. Comp. 22(2), 260–263]. Advances in virtual reality (VR) and avatar technologies have created new platforms for face-to-face communication in which visual speech information is presented through avatars. The avatars' articulatory movements may be generated in real time based on an algorithmic response to acoustic parameters. While the communicative experience in VR has become increasingly realistic, the visual speech articulations remain intentionally imperfect and focused on synchrony to avoid uncanny valley effects [https://developers.facebook.com/videos/f8-2017/the-making-of-facebook-spaces/]. Depending on the VR platform, vowel rounding may be represented reasonably faithfully while mouth opening size may convey gross variation in amplitude. It is unknown whether and how perceivers make use of such underspecified and at times misleading visual cues to speech. The current study investigates whether reliable segmental information can be extracted from visual speech algorithmically generated through a popular VR platform. We report on an experiment using a speech in noise task with audiovisual stimuli in two conditions (with articulatory movement and without) to see whether the visual information improves or degrades identification.

2pSC12. Single-channel vibrotactile feedback for voicing enhancement in trained and untrained perceivers. David G. Marino (Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), Hannah Elbaggari, Tzu Hsu Chu (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada, h.elbaggari@alumn.ubc.ca), Karon MacLean (Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Auditory speech intelligibility can be enhanced by integrating information from other modalities, e.g., vision [Sumby & Pollack 1954, *J. Acoust. Soc. Am.* 26: 212] or direct manual touch [Gick *et al.* 2008. *J. Acoust. Soc. Am.* 123: EL72]. There are, nonetheless, many circumstances where shared visual attention may be hard to establish, or where in-person contact may be infeasible (e.g., in a noisy collaborative environment). To test the feasibility of using vibrotactile feedback to enhance intelligibility under noisy conditions, we use a portable voice-coil-based transducer that provides vibrotactile stimulation similar to laryngeal vibrations. Participants were asked to discriminate between minimal pairs in noise. These were distinguished in voicing and vowel height. Participants were asked to wear a vibrator on their fingers, or on their suprasternal notch. We contrasted vibrator placement with different vibration styles, such as a constant vibration on voicing, or vibrations driven by the amplitude envelope of the speech signal. In untrained perceivers we found that vibrotactile feedback increased accuracy regardless of placement. This effect, though significant, was not strong enough to be useful for everyday speech enhancement. These results, and those of a follow-up study with trained perceivers, will be reported. [Funding from NSERC.]

2pSC13. Testing symmetry of temporal window of integration between vibrotactile and auditory speech information in voiced phoneme perception. Tzu Hsu Chu (Dept. of Linguist, Univ. of Br. Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, chusophy@alumni.ubc.ca), David G. Marino (Dept. of Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), Hannah Elbaggari (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), Karon MacLean (Dept. of Comput. Sci., Univ. of Br. Columbia, Vancouver, BC, Canada), and Bryan Gick (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

The temporal window for enhancement in cross-modal integration is asymmetrical, and does not require synchrony of stimuli for integration; this temporal offset has been linked to differences in relative signal speed as can be seen in audio-visual integration [Munhall *et al.* 1996. *Perc. Psychophys.* 58: 351] and audio-aerotactile integration [Gick *et al.* 2010. *Journ. Acoust. Soc. Am.* 128: EL342] of speech. However, as vibrotactile cues normally accompany acoustic cues, no difference in signal speed—and no concomitant perceptual asymmetry—is experienced between these two modalities. Contrary to previous research, we predict a symmetrical window for integration. A portable voice-coil transducer is used to produce vibrotactile stimuli, similar to the laryngeal vibrations normally felt in voiced speech. Results of an experiment will be presented in which participants are asked to discriminate between minimal pairs in noise with the device between their thumb and index finger, and in which audio and vibrotactile stimuli are presented in different orders and at different temporal offsets. Implications will be discussed for theories of cross-modal integration. [Funding from NSERC.]

2pSC14. The role of speaker identification in auditory and visual speech recognition: Evidence from twin speech. Madeline Petrich, Macie Petrich, Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover W225, Athens, OH 45701, leec1@ohio.edu), Seth Wiener (Modern Lang., Carnegie Mellon Univ., Pittsburgh, PA), Margaret Harrison, and Sarah Greenlee (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

Understanding the content of speech does not require identifying speaker voices. Similarly, identifying speaker voices does not require understanding the content of speech. However, ample evidence has shown that speaker voices affect auditory word recognition. Familiarity with speaker voices also affects audiovisual speech recognition. This study examines whether familiarity with twin speakers, who have highly similar voices, affects the magnitude of form priming and that of the McGurk effect. In Experiment 1, participants who were familiar or unfamiliar with the twin speakers performed lexical decision and voice discrimination on pairs of auditory words that were repeated or unrelated. In Experiment 2, the same participants were asked to identify syllables in which auditory and visual information was congruent or incongruent (e.g., auditory /ba/ with visual /ga/). The auditory and visual information was also mixed between the speakers. Experiment 1 showed that familiarity resulted in reduced priming in voice discrimination but not lexical decision. Experiment 2 showed that familiarity did not result in reduced McGurk effect. It appears that the benefit of familiarity in processing auditory and visual speech is compromised when processing highly similar voices. [Work supported by Ohio University PURF and CHSP Student Research Grant.]

2pSC15. Auditory lexical decision in the wild. Benjamin V. Tucker, Filip Nenadic, and Matthew C. Kelley (Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

The present report describes a version of the MALD (Massive Auditory Lexical Decision) database which investigates listener performance in the

often crowded TELUS World of Science—Edmonton (TWOSE) on tablets with headphones, with most participants experiencing varying levels of uncontrolled distraction. A total of 533 listeners participated in the experiment (ages 4–86; 51% female; 81.8% native English). Stimuli were a randomly selected subset of 2000 words and 2000 pseudowords extracted from the MALD project (Tucker *et al*., 2018), recorded by a single speaker. The stimuli were further randomly divided into 20 lists, each containing 100 words and 100 pseudowords, with no practice stimuli. Each participant was presented with a single list, with the entire experimental session usually lasting between five and ten minutes. TWOSE-MALD participants, unsurprisingly, perform worse than listeners in a laboratory setting in terms of both accuracy and response latencies. The relationship between the standard predictors and response latency remains the same. We also find age related effects indicating that accuracy increases rapidly (from ages 4 to 14) and slowly plateaus, while response latencies rapidly decrease until participants reach their early twenties, after which a steady increase is noted as participants' age increases.

2pSC16. The time course of recognition of reduced disyllabic Japanese words: Evidence from pupillometry with a Go-NoGo task. Yoichi Mukai, Benjamin V. Tucker, and Juhani Järvikivi (Dept. of Linguist, Univ. of AB, 3-26 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, mukai@ualberta.ca)

While much attention has been paid to the importance of reduction in spoken word recognition, fewer studies have investigated the effect of reduction over time. Thirty-eight participants' pupillary responses were measured during the perception of Japanese disyllabic words as they performed a Go-NoGo task. We used 226 lexical items, each of which contained both reduced and citation forms of the words. All stimuli consisted of a word-medial nasal or voiced stop. Results demonstrate that the overall amount of cognitive effort required to process reduced forms was higher than that of canonical forms. That is, greater pupil dilation was observed for reduced forms than for citation forms. This result is in line with previous research (e.g., Tucker, 2011). Specifically, pupil dilation was greater with reduced forms in the time window of 436 ms to 2000 ms after the onset of stimuli. Our results also indicate that the pattern of pupil dilation over time with reduced forms differs from citation forms, indicating that reduced forms show a later onset and offset of peak dilation and more gradual constriction of pupil compared to citation forms.

2pSC17. Multimodal recognition of interrupted speech: Benefit from text and visual speech cues. Rachel E. Miller, Courtney Strickland, and Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1229 Marion St., Columbia, SC 29201, remiller@email.sc.edu)

Presenting degraded speech with visual cues facilitates speech recognition. This benefit is observed for visual speech cues that are perceptually correlated with the auditory signal, as well as for text cues that delay integration until a later cognitive-linguistic processing stage. However, it is not clear how the benefit compares between these two types of degraded multimodal presentations. The current study examined how listeners integrate visually interrupted text or visual speech cues with acoustically interrupted speech. In Experiment 1, text was periodically interrupted by white space at visual interruption rates that were associated with the auditory interruption rate of speech. In Experiment 2, videos were visually interrupted by grey frames. The synchrony of audio-visual interruption was also manipulated by presenting visual cues in-phase or 180^{o} out-of-phase with speech interruptions. For both experiments, speech was low-pass filtered at 2000 Hz. Preliminary results indicate that listeners obtain a benefit from both visual speech and text cues. In addition, performance is affected by the interruption rate of speech, with minimal performance obtained around an interruption rate of 2 Hz. Supplementing speech with incomplete visual cues can improve sentence intelligibility and compensate for degraded speech in adverse listening conditions. [Work supported, in part, by NIH/NIDCD.]

2pSC18. Phonetic recalibration of visual and auditory speech by visual sentences. Josh Dorsi (Psych., UC Riverside, 34875, Winchester, NY 92596, jdors002@ucr.edu), Lawrence Rosenblum, and Sharon Chee (Psych., UC Riverside, Riverside, CA)

Phonetic recalibration is a form of perceptual learning in which experience with an ambiguous speech segment accompanied by disambiguating context biases subsequent perception of that segment. For example, presenting an auditory "f" final word (i.e., "Witlof"; Dutch for "Chicory") with the final fricative replaced with a sound between /s/ and /f/ results in participants subsequently categorizing more items from a /f/-/s/ continuum as /f/ (Norris, McQueen, Cutler 2003). A similar effect is found for ambiguous visual speech that is accompanied by clear auditory speech (Baart & Vroomen 2010), and for ambiguous auditory speech that is accompanied by clear visual speech (e.g., Vroomen & Baart, 2009). It is unknown whether visual speech alone can recalibrate visual speech segments or whether recalibration in one modality can be transferred to another modality (but see, Dias & Rosenblum 2016). In Experiment 1 of this study we found that sentence context disambiguated a silent visual ambiguous /s/f/ fricative and resulted in visual recalibration. Experiment 2 used these same silent sentences but measured recalibration on an auditory continuum. Preliminary results suggest that recalibration effects can be transferred across modalities in this way. These experiments have implications for theories of amodal perceptual learning.

2pSC19. Lexical access in the face of degraded speech: The effects of cognitive adaptation. Francis Smith and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, W311 Seashore Hall, Iowa City, IA 52245, francis-smith@uiowa.edu)

Hearing impaired listeners use cognitive adaptations to cope with degraded input. Here, we ask if they adapt processes that normal hearing listeners use to cope with the fact that speech unfolds over time, creating brief periods of ambiguity. Normal listeners cope with this ambiguity by activating multiple lexical candidates which compete for recognition (McClelland & Elman, 1986). These competition dynamics change when processing degraded input (Brouwer & Bradlow, 2016; McMurray, Farris-Trimble, & Rigler, 2017; McQueen & Huettig, 2012), but it is unclear whether this reflects the degraded input itself or cognitive adaptation. In two visual world paradigm experiments, listeners heard different levels of degraded (noise-vocoded) speech. Experiment 1 manipulated the level of degradation either in blocks or randomly interleaved across trials. Interleaving led to processing delays beyond that of the level of degradation: we found switch-costs when degradation levels differed between trials. This suggests differences in lexical dynamics are not solely due to degradation in the input. In Experiment 2, a visual cue indicated the level of degradation before each trial. This reduced the processing delays and switch costs, suggesting participants adapted before the auditory input. These experiments support a role for central processing in dealing with degraded speech input.

2pSC20. Integration of speech information (or not) across electric and acoustic modes in hearing impaired listeners. Michael Seedorff (Biostatistics, Univ. of Iowa, Iowa City, IA) and Bob McMurray (Psych., Univ. of Iowa, E11 SSH, Iowa City, IA 52242, bob-mcmurray@uiowa.edu)

Many people with hearing loss use two cochlear implants (bilateral CIs), or combine a CI and hearing aid (bimodal). While the addition of a second CI or acoustic input improves speech perception (particularly in noise), it is not well understood how listeners fuse these disparate inputs. We addressed this with a duplex perception experiment. Listeners heard the four cardinal vowels from six talkers. In the duplex condition, the first formant was presented to the CI, and the second to the other CI or hearing aid (and vice versa). In normal listeners ($N = 14$), accuracy under duplex presentation (across ears; $M = 0.80$) did not differ from combined presentation (both formants, both ears, $M = 0.76$, $p = 0.14$), but performance dropped to near chance with isolated formants ($M = 0.40$, $p < 0.001$). CI users (Bilateral: $N = 7$; bimodal: $N = 11$; single-side-of-deafness [SSD]: $N = 7$) performed well in combined presentation (Bilateral: $M = 0.72$, Bimodal: $M = 0.70$, SSD: $M = 0.83$), and poorly for isolated formants (Bilateral: $M = 0.42$,

Bimodal: M = 0.39, SSD: M = 0.41). Duplex presentation was significantly poorer than combined (Bilateral: M = 0.54, p = 0.0078; Bimodal: M = 0.55, p<0.001; SSD: Duplex: 0.48, p<0.001), though better than isolated formants (p<0.05). Thus, CI users may not fuse inputs well at an auditory level, and the benefits of acoustic + electric hearing may derive from central integration.

2pSC21. The effect of perceptual similarity, frequency, and phonotactic restriction in loanword adaptation. Yang-Yu Chen (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, Hsinchu, Taiwan, Taiwan) and Yu-an Lu (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, F319, Humanities Bldg. 2, 1001 University Rd., Hsinchu 30010, Taiwan, yuanlu@nctu.edu.tw)

Mandarin speakers tend to adapt an intervocalic nasal as either an onset of the following syllable (e.g., *Bruno*à *bù.**lǔ**.nuò*) or as a nasal geminate (e.g., *Daniel* à *d**ān.n**í.ěr*) (Huang & Lin 2013, 2014). Two forced-choice identification experiments were conducted to test the effects of nasalization (whether the pre-nasal vowel bears stress or not) and duration (whether the pre-nasal vowel is lax or tense): Would stronger nasalization and shorter duration encourage higher nasal gemination rate? The results showed that Mandarin speakers' choice of repairs was indeed biased by the different phonetic manipulations, suggesting an effect of perceptual similarity. Moreover, the overall preference for the V.NV form over the VN.NV form suggests an influence from the native syllable type frequency (open syllables being more frequent than closed syllables). The across-the-board higher VN.NV responses for lax than for tense vowels regardless of the phonetic manipulations are attributed to the possibility that Mandarin speakers might have perceived the tense vowels as diphthongs (i.e., English /e/ to [ej], /o/ to [ow]) and inserting a nasal coda is illegal in this contexts (*CVGN). That is, the findings suggest that the variations in loanword adaptation were guided by perception, frequency, as well as phonotactics.

2pSC22. Effects of delay time and intensity difference on recognition of overlapped speech sound. Shigeaki Amano (Faculty of Human Informatics, Aichi Shukutoku Univ., 9 Katahira, Nagakute, Aichi 480-1197, Japan, psy@asu.aasa.ac.jp), Kimiko Yamakawa (Shokei Univ., Kikuchi-gun, Kumamoto, Japan), and Katuhiro Maki (Faculty of Human Informatics, Aichi Shukutoku Univ., Aichi, Japan)

Speech sound transmitted from an outdoor loudspeaker is sometimes difficult to recognize because it is overlapped by a time-delayed speech sound from other loudspeakers. This difficulty is assumed to depend on a delay time and an intensity difference between the original and overlapping time-delayed sounds. To clarify the effects of a delay time and an intensity difference on speech recognition, a listening experiment was conducted with 21 Japanese adults using 105 Japanese spoken words in a carrier sentence, with delay times and intensity differences ranging between 0–250 ms and 0–9 dB, respectively. The experiment revealed that recognition ratios are significantly lower for delay times of 100–250 ms than for 0 ms. Similarly, the ratios are significantly lower for intensity differences of 0–6 dB than for 9 dB. These results suggest that speech sound from an outdoor loudspeaker is difficult to recognize in a large area where the difference of distances from two loudspeakers ranges between 34 and 85 m and the intensity difference between the original and overlapping time-delayed sound is less than 6 dB. [This study was supported by JSPS KAKENHI Grant Numbers JP15K12494, JP15H03207, JP17K02705, and by Aichi-Shukutoku University Cooperative Research Grant 2017-2018.]

2pSC23. Different musical experiences differentially modulate the auditory-motor processing of feedback errors during vocal production. Wenda Wang and Hanjun Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

The present study investigated the effects of musical experiences on auditory-motor control of pitch feedback errors during vocal production. Thirty-four female musicians who were assigned to a group of professional signers (n = 17) and a group of instrument players (n = 17) and seventeen female non-musicians aged 18–29 years participated in the present study. All participants vocalized a vowel sound /u/ for about 5-6 seconds while hearing their vocal pitch unexpectedly shifted -50 or -200 cents, and their vocal compensations for pitch perturbations were measured and compared. The results showed significantly larger compensatory vocal responses produced by non-musicians when compared to professional singers (p<0.001) and instrument players (p<0.008). Moreover, instrument players produced significantly larger vocal compensations than professional singers (p = 0.026). The observed effects of musical experiences were independent of the size of pitch perturbation. These findings provide the first behavioral evidence for the differential modulation of vocal compensations for pitch perturbations by different musical experiences, which may be related to the difference between singers and players in extensive vocal training.

2pSC24. Influence of word size and tonal sequence probabilities on Mandarin segmentation errors. Amy LaCross (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., PO Box 870102, Tempe, AZ 85287, amy.lacross@asu.edu), Jordan Sandoval (Linguist, Western Washington Univ., Bellingham, WA), and Julie Liss (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

The largely effortless process of segmenting a continuous speech stream into words has been shown cross-linguistically to be influenced by implicit knowledge about the distributional probabilities of phonotactics, stress assignment, and word size. Work in tonal languages provides compelling evidence that tones are also likely to be a source of probabilistic information for speakers of these languages. It has been shown that probability of syllable + tone combinations play a role in speech processing in Mandarin (Wiener & Ito, 2016; 2015; Wiener & Turnbull, 2013). Work in Cantonese has also demonstrated that transitional probabilities of lexical tones paired with vowels aid listeners in segmentation (Gomez, *et al*., 2018). Here we conducted two perception experiments with fifty native-Mandarin speakers and manipulated two potential segmentation cues: word size and tonal sequence probability. Contrary to our hypothesis that participants would make segmentation errors which reflect the most probable word size in Mandarin (two-syllable words) with highly probable tonal sequences, participants' errors were overwhelmingly three-syllable words with low probability tonal sequences. This suggests that biases towards segmentation errors are not predictable based on straightforward probabilities.

2pSC25. Concurrent aero-tactile stimulation does not bias perception of VOT for non-initial stops. Dolly Goldenberg (Linguist, Yale Univ., 192 Foster St., Apt 1, New Haven, CT 06511, dolly.goldenberg@yale.edu), Mark Tiede, and D. H. Whalen (Haskins Labs., New Haven, CT)

Previous work has established that puffs of air applied to the skin and timed with listening tasks bias the perception of voicing in onset stops by naive listeners (Gick and Derrick, 2009; Goldenberg *et al*., 2015). While the primary cue for the voicing contrast in stops is VOT (Lisker and Abramson, 1964), in English aspiration typically functions as a cue foot initially. This study tests the effect of air puffs on perception of voicing for English stops in a non-foot-initial context ("apa/aba") using VOT continua. Goldenberg *et al*. (2015) have shown that listeners are sensitive to aero-tactile effects only when these are congruent with the expected contrast (i.e., in VOT but not vowel quality distinctions). Since VOT is generally non-contrastive for English stops that are not foot-initial, air puffs were not expected to affect perception in the current case, and indeed, of 22 participants (11 females; mean age 34.2) tested, 20 showed no effect. Comparison of this null result to the significant bias observed in the earlier (foot-initial context) study extends the finding that, for aero-tactile stimulation to bias perception, the cues must be consistent with those expected in production of the perceived sounds.

2pSC26. Bimodal classification of English allophones employing acoustic speech signal and facial motion capture. Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Szymon Zaporowski, and Bozena Kostek (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland)

A method for automatic transcription of English speech into International Phonetic Alphabet (IPA) system is developed and studied. The principal objective of the study is to evaluate to what extent the visual data related

to lip reading can enhance recognition accuracy of the transcription of English consonantal and vocalic allophones. To this end, motion capture markers were placed on the faces of seven speakers to obtain lip tracking data synchronized with the audio signal. 32 markers were used, 20 of which were placed on the speaker's inner lips and 4 on a special cap, which served as the point of reference and stabilized the FMC image while post-processing. Speech samples were simultaneously recorded as a list of approximately 300 words in which all English consonantal and vocalic allophones were represented. Different parameterization strategies were tested and the accuracy of vocalic segments recognition in different experiments was analyzed. The process of multimodal feature extraction is explained, the applied feature selection methods are presented and the obtained results are discussed. Further challenges related to the bi-modal feature extraction process and neural network-based decision systems employment are discussed. [Research sponsored by the Polish National Science Centre, Dec. No. 2015/17/B/ST6/01874.]

2pSC27. Effects of spatiotemporal manipulation of audiovisual speech on the perception of prosodic structure. Robert Fuhrman (Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, robert.fuhrman@alumni.ubc.ca)

Work in audiovisual speech processing (AVSP) has established that the availability of visual speech signals can influence auditory perception by improving the intelligibility of speech in noise (Sumby and Pollack, 1954). However, exactly which aspects of visible signals are most responsible for this enhancement remains an open question, although convergent evidence along several lines suggests that visible information may reflect a common articulatory-acoustic temporal signature, and that the multi-modal availability of this temporal signature is at the root of this effect. We evaluated this hypothesis in a perceptual study using simple talking face animations whose motion is driven by a signal derived from the collective motion of perioral structures of an actual talker. We applied spatial and temporal manipulations to the structure of this driving signal using a biologically plausible model that preserves the smoothness of the manipulated trajectory, and tested whether these kinematic manipulations influenced the perception of linguistic prominence, an important component of the timing and rhythm (prosody) of speech. The data suggest that perceivers are sensitive to these manipulations, and that the cross-correlation between the acoustic amplitude envelope and the manipulated visible signal was a strong predictor of the perception of prominence.

2pSC28. Integration of literal meaning of emotional phrases with vocal emotion: Comparison between Japanese and North Americans. Sumi Shigeno (Psych., Aoyama Gakuin Univ., 4-4-25 Shibuya, Shibuya-ku, Tokyo 150-8366, Japan, sshigeno@ephs.aoyama.ac.jp)

This study investigated how the display rules of emotional expression apply in vocal-only communication depending on speakers' and listeners' cultural backgrounds and their similarities and differences. A comparison was made between Japanese and North American participants, who listened to emotional phrases spoken in their native language and a non-native language (English and Japanese, respectively). The participants were instructed to listen carefully to the recordings and judge the speakers' true emotions. The speakers had been asked to express an emotion with their voice that was either congruent or incongruent with the emotion contained in the literal meaning of speech (ELMS), which, in English, corresponded to: "eleven-thirty," "good afternoon," "congratulations," "I love it," "I'm going to cry," and "my heart is breaking." The speakers spoke these utterances in neutral, happy, and sad voices. The results indicated that the Japanese listeners integrated the ELMS with the vocal emotions when the speakers were Japanese but judged the speakers' emotions based on voice alone when the speakers were North American. This implied that Japanese participants could infer the speakers' true emotions even when the ELMS was incongruent with vocal emotion. The results obtained from the Japanese and North American speakers and listeners were compared.

2pSC29. Acoustic features of intelligible speech produced under reverberant environments. Rieko Kubo and Masato Akagi (JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, rkubo@jaist.ac.jp)

Clear speech has been shown to have an intelligibility advantage over conversational speech in reverberant environments. Speech produced in reverberant environments is also reported to be more intelligible than ones produced in normal environments. Speakers adapt to the environments by modifying their articulation, and acoustics of speech. The speech produced under reverberant conditions should be modified similarly with the clear speech. Identifying the distinctive features which are shared between these speech types gives insights into which acoustic features are robust cues even in reverberant environments. In the present study, we investigated voices uttered by speakers who produced Japanese words under reverberant conditions (T60 = 1 s–5 s). Speech intelligibility tests, and acoustic analyses were performed. The results showed that, first, speech produced under long T60 is more intelligible in reverberant environments. Second, the intelligible speech has expanded vowel space with high F2 for front vowels and low F2 for back vowels, and high F1 for open vowels and low F1 for close vowels. Brief formant transitions were also found. These articulatory/acoustic modifications are likely to be enhancements of significant perceptual cues. The acoustic/articulatory features of each vowel become clearer (hyper-articulated speech; i.e., clear speech.) Moreover, perceptual compensation to reduce co-articulation effects should become easier.

2pSC30. Speech intelligibility testing of general service respirators. Christoph Hoeller, Markus Müller-Trapet, and John S. Bradley (Construction, National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, christoph.hoeller@nrc.ca)

The communication performance of three general service respirators was evaluated. The evaluation procedure followed a NIOSH standard test procedure which is based on a modified rhyme test (MRT), wherein human participants are placed in a controlled acoustic environment and are asked to read out lists of test words. The number of words correctly identified by a listener panel when the talker is wearing the respirator is compared to the number of words correctly identified without the respirator, giving an overall performance rating. During the implementation of the experimental procedure, it was noted that the NIOSH test procedure is ambiguous on a number of potentially significant acoustical issues, e.g., the reverberation time of the test environment, the background noise spectrum at the listener position, and the talker voice level. This talk will present details of the study and discuss the relevant issues related to speech intelligibility testing of respirators that were evaluated as part of this project.

2pSC31. Measuring the effect of speaker ethnicity on online perception: Evidence from a response time study. Noortje de Weers (Linguist, Simon Fraser Univ., 316- 4453 Main St., Vancouver, BC V5V0A2, Canada, ndeweers@sfu.ca)

The effect of speaker ethnicity on speech perception remains unclear. Proponents of the bias hypothesis maintain that presenting an Asian or Mexican face to American participants triggers a certain kind of bias that could result in worse comprehension and even hearing a non-existent 'foreign accent.' Exemplar-based studies, on the other hand, have proposed that these findings merely reflect a mismatch between listeners' expectations and the actual speech signal. While previous studies all used post-perceptual, offline tasks to examine the effect of speaker ethnicity on speech perception, this study made use of an online task instead. Thirty-two native English participants completed a speeded audio-visual sentence verification task, for which they had to classify statements as true or false. The utterances were paired with a photograph of an Asian face, a White face, or a fixation cross, and were presented in a mixed design. Both correctness scores and response times for all the different face-voice pairings were recorded. Results suggest that online processing was not affected by speaker ethnicity, as response times did not differ as a function of the various face-voice pairings. Additional findings showed that the foreign-accented voices took significantly longer to process than the native voices, and that false statements took longer to answer than true statements.

2pSC32. Perception of sexual orientation through speech: Generational differences. Lily Obeda, Melanie Putman, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Previous research has shown that listeners can perceive talkers' sexual orientation at a greater than chance levels from the acoustic characteristics of their speech (Munson *et al*., J. Phonetics [2006]). Recently, Obeda and Munson (2018, J. Acoust. Soc. Am. 143, 1924) compared the perception of 44 gay, lesbian, and heterosexual people's sexual orientation through speech by the young adults reported in Munson *et al.* 2006 (whose data were collected in 2003) to a group of young adults tested in 2018. The new listeners were matched in age to the 2003 listeners. A number of differences between the two cohorts were found. In particular, the 2018 listeners were less willing to judge heterosexual women's speech as 'sounding heterosexual'. In this presentation, we compare both sets of data to the performance of a new set of listeners who are matched in birth year to the 2003 listeners, and who are currently 33–48 years old. Collection of those data is ongoing. Comparing this new group to the two other groups will allow us to examine whether the apparent generational change observed by Obeda and Munson has led listeners like those in 2003 to reevaluate their perception of sexual orientation through speech in 2018.

2pSC33. Regional dialect and talker sex information in high-frequency region. Robert A. Fox, Ewa Jacewicz, and Magan McClurg (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

High-pass filtering, in which the higher frequencies are retained and the lower frequencies are eliminated, has not been used in speech perception research as often as low-pass filtering because most intelligibility cues have been assumed to reside in the low-frequency region, up to 4 kHz (French and Steinberg, 1947). Recent research provided new evidence that listeners can also utilize high-frequency energy in the perception of voice, talker sex, and speech characteristics such as naturalness, pleasantness, and clarity, which improves speaker and word recognition in noise. Given the potential for the existence of accessible linguistic and paralinguistic information, the current study asks whether perception of regional dialect can be influenced by spectral content in the high-frequency region, and how robust this information is—in males and females—when low-frequency cues are unavailable. Listeners from Ohio heard phrases produced by males and females from Ohio and Western North Carolina high-pass filtered at 700, 1175, 1973, 3312, and 5560 Hz. Each higher filter provided increasingly fewer cues about dialect whereas sex identification remained relatively high. Female speech provided significantly more dialect cues than male speech when more spectral information was available (filter cut-offs at 1175 and 1973 Hz, but not at 700 Hz).

2pSC34. Distinguishing Dick from Jane: Children's voices are more difficult to identify than adults' voices. Natalie Fecher, Angela Cooper, and Elizabeth Johnson (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5G4K2, Canada, angela.cooper@utoronto.ca)

Previous research on talker recognition has focused on the processing of talker identity in adult voices; however, little is known about how listeners identify child voices. This study examined adult listeners' ability to differentiate and identify child voices (2.5-year-olds) and adult voices. In Experiment 1, native English-speaking listeners completed an AX voice discrimination task with both child and adult voices. Results revealed that listeners were significantly worse at discriminating between child voices relative to adult voices, regardless of the fact that child voices were mixed gender and adult voices were all female. In Experiment 2, we examined whether listeners could learn to identify child and adult voices. Adult listeners completed an adaptive training paradigm, where they learned to identify 4 child voices on one day and 4 adult voices on a separate day. Preliminary results indicate that with training, listeners can learn to identify voices from both age groups above chance. However, listeners were still faster at learning and more accurate at identifying adult relative to child voices. Taken together, these findings are an initial step towards understanding how talker recognition is influenced by the acoustic-phonetic characteristics and articulatory capabilities that vary with talker age.

2pSC35. Visual gender biases in English stop voicing perception. Connie Ting, Rachel Soo (Univ. of Toronto, 100 St. George St., Toronto, ON, Canada, connie.ting.ct@gmail.com), and Jessamyn L. Schertz (Univ. of Toronto, Mississauga, ON, Canada)

Listeners leverage visual information, such as perceived speaker gender, during speech perception. While work has shown visual gender biases on fricative (Johnson & Strand, 1996) and vowel perception (Johnson *et al*., 1999), its effects on stop voicing perception are understudied. We present an identification task where visual gender primes (male/female portraits) were used to investigate perceived speaker gender effects on English stop voicing. Subjects ($n = 22$) identified 'pa'/'ba' syllables from an 11-step VOT continuum (0–50 ms) in 5 pitch levels (high: 250 Hz, 230 Hz, mid: 170 Hz, and low: 130 Hz, 110 Hz). High and low pitch tokens were always preceded by a female and male portrait, respectively. Mid pitch tokens were preceded by female or male images in separate blocks. Given that f0 is higher following voiceless obstruents (Lehiste & Peterson 1961; Mohr 1971), if listeners utilize visual gender cues, acoustically ambiguous tokens paired with a male portrait should elicit more voiceless "pa" responses than those paired with a female portrait. Preliminary results pattern in the expected direction; mid tokens are perceived as "pa" more when paired with a male (15–25 ms = 79% "pa" response) than a female face (73% "pa" response); however, this effect was not statistically significant.

2pSC36. Effects of emotional tones of voice on the acoustic and perceptual characteristics of Mandarin tones. Huishan Chang (Dept. of Audiol. and Speech-Lang. Pathol., Asia Univ., No. 500, Liufeng Rd., Wufeng Dist., Taichung 41354, Taiwan, sandychang0217@gmail.com), Shuenn-Tsong Young (Holistic Education Ctr., Mackay Medical College, New Taipei, Taiwan), Pei-Chun Li (Dept. of Audiol. and Speech-Lang. Pathol., Mackay Medical College, New Taipei, Taiwan), Woei-Chyn Chu (Dept. of Biomedical Eng., School of Biomedical Sci. and Eng., National Yang-Ming Univ., Taipei, Taiwan), and Cheng-Yu Ho (Holistic Education Ctr., Mackay Medical College, New Taipei, Taiwan)

Emotional tones of a speaker's voice and lexical tones involve similar acoustic correlates, but only lexical tones could change the meaning of a word in tonal languages. The purpose of this study is to investigate the interaction between these two types of tonal variations by examining the acoustic and perceptual characteristics of the four Mandarin tones across different emotional tones of voice. In experiment 1, acoustic analyses of fundamental frequency, mean amplitude, and duration was conducted on a syllable with the four tones produced in a carrier phrase with four different emotional tones of voice (anger, fear, happiness, and sadness). The same acoustic measures were also taken on the Mandarin neutral tone produced with the four emotional tones. In experiment 2, speech materials from experiment 1 were used to investigate the effects of the emotional tones on the perception of Mandarin tones. The results showed that all four emotional tones had significant effects on the acoustics and perception of Mandarin tones. These findings suggest that emotional tones of voice impact both acoustic and perceptual characteristics of lexical tones.

2pSC37. Acoustic emotion recognition using spectral and temporal features. Tejal Udhan (Dept. of Elec. and Comput. Eng., Florida State Univ., 2525 Pottsdamer St., Tallahassee, FL 32310, tu13b@my.fsu.edu) and Shonda Bernadin (Dept. of Elec. and Comput. Eng., FAMU- FSU College of Eng., Tallahassee, FL)

In this paper, utility of different low- level, spectral and temporal features is evaluated for the task of emotion recognition. The aim of an ideal speech emotion recognition system is to extract features that are representative of the emotional state of speaker. Pitch, intensity, frequency formants, jitter, and zero crossing rate are five features proposed for characterizing four different emotions, anger, happy, sadness, and neutral. Low- level spectral and temporal features have ease of calculation and limit the complexity of emotion recognition systems since they are commonly single dimensional features. A decision-tree based algorithm is designed for characterizing emotions using these acoustic features. It has been proven that various aspects of a speaker's physical and emotional state can be identified by speech alone. However, the accuracy of such analyses has not been optimized due to acoustic variabilities such as length and complexity of human speech utterance, gender, speaking styles, and speech rate. Since speech emotion recognition is a developing and challenging field, most powerful features for emotion recognition are not yet defined; hence, investigating the utility of selected features for emotion recognition is an important task.

2pSC38. A methodological framework to derive the mental code of emotional and social inferences in sound. Emmanuel Ponsot (ENS, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

Most research carried out so far in auditory emotion has been based on theory-driven experiments. Such experiments, which involve a limited number of conditions, help in refining existing models or theories but cannot expose elements or processes that were not expected and targeted initially. Here, I will present a methodological framework based on a data-driven approach that allows to probe the mechanisms of auditory cognition in a space that is not constrained by a priori hypotheses. This framework uses signal-processing algorithms for manipulating the acoustical characteristics of an input sound and create a huge number of parametrically-manipulated, natural expressive variations of this sound, which are then used as stimuli in psychophysical experiments employing a reverse-correlation technique. I will present different contexts in which this framework has been used, in particular to explore the processing of speech prosodic dimensions in the formation of social impressions. Because this approach offers a principled way to reverse-engineer any high-level judgment in any individual, it should be helpful to understand the algorithms that the brain uses to build high-level emotional or social impressions from the acoustical characteristics of sound in their complexity.

2pSC39. Study on the relationship between modulation spectral features and the perception of vocal emotion with noise-vocoded speech. Zhi Zhu, Ryota Miyauchi (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., 5-201 Student House, 1-1 Asahidai, Nomi, Ishikawa 9231211, Japan, zhuzhi@jaist.ac.jp), Yukiko Araki (Kanazawa Univ., Kanazawa, Ishikawa, Japan), and Masashi Unoki (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi-shi, Japan)

Previous studies about vocal-emotion recognition with noise-vocoded speech showed that temporal modulation cues provided by the temporal envelope play an important role in the perception of vocal emotion. To clarify the exact feature of temporal envelope that contributes to the perception of vocal emotion, a method based on the mechanism of modulation frequency analysis in the auditory system is necessary. In this study, auditory-based modulation spectral features were used to account for the perceptual data collected from vocal-emotion recognition experiments using noise-vocoded speech. At first, the modulation spectrogram of the emotional noise-vocoded speech was calculated by using an auditory-based modulation filterbank. Then, ten types of modulation spectral features were extracted from the modulation spectrograms. Finally, modulation spectral features and the perceptual data were compared to investigate the contribution of temporal envelope to the perception of vocal emotion with noise-vocoded speech. The results showed that there were high correlations between modulation spectral features and the perceptual data. Therefore, the modulation spectral features should be useful for accounting for the perceptual processing of vocal emotion with noise-vocoded speech. [Work supported by JSPS KAKENHI Grant Number JP. 17J08312, and Grant in Aid for Scientific Research Innovative Areas (No. 18H05004) from MEXT, Japan.]

2pSC40. Proposals of noise-robust spoken words for a broadcast via outdoor loudspeakers. Kimiko Yamakawa (Cultural Commun., Shokei Univ., 2-8-1 Musashigaoka-kita, Kikuyo, Kikuchi-gun, Kumamoto 8618538, Japan, jin@shokei-gakuen.ac.jp) and Shigeaki Amano (Aichi Shukutoku Univ., Nagakute, Aichi, Japan)

Spoken words used in a broadcast via outdoor loudspeakers were analyzed in terms of word frequency, word familiarity, and phoneme composition to clarify their characteristics. Based on the analysis, we identified 13 original words that are frequently used but probably difficult to listen to and proposed alternative words having higher word familiarity and a lesser number of noise-intolerant phonemes. To verify the proposal, a listening experiment was conducted. Thirteen each of the original and proposed words were presented with and without pink noise (SNR = 0 dB) to 21 participants with normal hearing ability. Error ratios of word identification were obtained from participants' responses. In a with-noise condition, the error ratio of the proposed words (7.1%) was significantly lower than that of the original words (18.3%), but no difference in error ratio was observed between the original and proposed words in a without-noise condition. These results indicate that the proposed words have higher noise robustness and less listening difficulty than the original words, and that they are useful for transmitting accurate information in a loudspeaker broadcast.

2pSC41. Optimal use of the visual analog scale: Observations from ratings of emotional speech. Shae D. Morgan (Univ. of Louisville, 390 South 1530 East, Ste. 1201, Salt Lake City, Utah 84112, shae.morgan@utah.edu)

Researchers use many tools to measure participant performance during experiments. The visual analog scale has been used in many areas of research as a valid way to measure listener perception of various aspects of a signal. Ratings using the visual analog scale are commonly averaged over a number of trials to estimate listeners' perception. This presentation delves into issues which may be encountered when using a visual analog scale for measuring perception of voice (in this case, vocal emotion). An experiment was performed which gathered ratings of emotional activation and pleasantness for many stimuli of four emotion categories (angry, sad, happy, and calm) from 10 listeners. These data will be examined for patterns of variance in a time-course analysis to identify listener certainty or biases created when using the visual analog scale during a long experimental task. Implications of bias and the impact on research results will be discussed along with additional and alternative measurement options for researchers to consider.

2pSC42. It's time to collaborate: What human linguists can learn from machine linguists. Michael D. Fry (Linguist, Univ. of Br. Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, mdfry20@gmail.com)

For decades, one of the primary goals of machine learning has been to emulate human performance by learning from human behavioural data. However, machine learning has now matured to a point where humans are learning from machines—for example, consider playstyles of Go (Silver, D., *et al.,* "Mastering the game of go without human knowledge," *Nature* 550.7676(2017):354). Drawing on this idea, this project considers what speech scientists might be able to learn from considering how machines analyze the speech signal. One goal in phonetics is to break down the speech signal into units such as phonemes, syllables, and tones. Machines are also able to break down the speech signal in either a supervised or unsupervised manner. In supervised learning, the machine is trained to classify phonemes, tones, etc., using previously known labels. In unsupervised learning, the machine learns a lower-dimensional, latent representation of the speech signal and then identifies meaningful clusters in the latent space. This project takes an unsupervised approach to lexical tone identification in Mandarin. Specifically, an adversarial autoencoder and density-based clustering are used to identify the tones of Mandarin. Results contrast Mandarin lexical tones as determined by linguists and by machine learning.

Session 2pUWa

Underwater Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Unmanned Vehicles and Acoustics

Erin Fischell, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., WHOI, MS 11, Woods Hole, MA 02543

Chair's Introduction—1:00

Invited Papers

1:05

2pUWa1. Bearing and range tracking with a two-hydrophone ocean glider. Elizabeth T. Küsel and Martin Siderius (Portland State Univ., Ste. 160, 1900 SW 4th Ave., Portland, OR 97201, ekusel@pdx.edu)

A Slocum glider fitted with two hydrophones separated by a distance of approximately 3 feet recorded sperm whale sounds during an experiment off the west coast of the island of Sardinia, Italy, in the summer of 2014. Time difference of arrival analysis of their echolocation clicks provided bearing tracks. It was also found that the left-right ambiguity inherent of the bearing estimation process could be broken by the natural oscillation in glider heading. More recently, a controlled experiment on a quiet lake with a known fixed acoustic source was devised to investigate unambiguous bearing estimation and possible ranging. Simple target motion analysis (TMA) using the glider's position data resulted in range estimates of only a few hundred meters short of the true source location. Analysis of the acoustic data on the other hand, presented challenges to satisfactory TMA performance. Overall, it was found that simple bearing-only TMA analysis is possible with two-hydrophone acoustic data and small angular change of a few degrees in the heading of the observing platform. Results will be discussed with implications to marine mammal population density estimation studies, since the ability to estimate direction and range of sound-producing animals could improve estimation of detection probabilities.

1:25

2pUWa2. Relative navigation for command and control of multiple low-cost autonomous underwater vehicles. Nicholas R. Rypkema (Elec. Eng. and Comput. Sci., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-223, Cambridge, MA 02139, rypkema@mit.edu), Erin Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

A long-held goal of autonomous underwater vehicle (AUV) research has been the coordinated use of multiple vehicles for novel applications. Unfortunately, the underwater domain poses significant challenges for navigation and communication—sophisticated navigational sensors increase vehicle price, size, and power use, and acoustic communications are limited in bandwidth and channel capacity. These factors make multi-vehicle deployments rare and limited in scale, with operators commanding each AUV individually and with limited coordination. In this work, we describe an operating paradigm that enables user-friendly command and control of multiple AUVs. Each vehicle uses a custom low-power, low-cost acoustic system to navigate and receive operator commands, consisting of a passive hydrophone array and timed acquisition and processing boards which enable the vehicle to self-localize relative to a synchronized acoustic beacon. By selecting between various beacon-transmitted signals, all vehicles can be simultaneously commanded to switch between behaviors. Additionally, a consequence of beacon-relative navigation is that movement of the beacon results in the concurrent movement of all AUVs. This system is ideal for multi-vehicle operations using inexpensive, miniature AUVs, as it does not require conventional high-cost navigational sensors or acoustic modems, and its passive nature and associated operating scheme enable the deployment of an arbitrarily large number of AUVs. [Work supported by Battelle, ONR, Lincoln Laboratory, DARPA.]

1:45

2pUWa3. Incorporating real-time acoustic ranging and glider-based Doppler measurements to aid vehicle navigation. Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, swebster@apl.washington.edu), Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Andrey Shcherbina (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Aleksandr Aravkin (Dept. of Appl. Mathematics, Univ. of Washington, Seattle, WA), Craig M. Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

In the summer of 2017, two Seagliders equipped with low frequency acoustic recorders and 1 MHz acoustic Doppler current profilers (ADCPs) were deployed in the Canada Basin as part of a large-scale acoustic tomography experiment. Acoustic tomography sources were moored at approximately 175 m depth within an acoustic duct enabling acoustic transmission to be received at long ranges. The sources transmitted at a center frequency of approximately 250 Hz every four hours, and ranges between the sources and Seaglider were

estimated in real time using a WHOI MicroModem. These ranges can be incorporated into an extended Kalman filter for navigation. In addition, glider-mounted upward-looking ADCPs recorded ~26.5 m shear profiles every 15 seconds. These overlapping profiles can be used to estimate (in post-processing) both the local current profile on a per-dive basis and the glider's relative velocity through the water. In real time, the close-range ADCP velocity measurements can be used to estimate the glider's relative (Through The Water, TTW) velocity to improve the glider's subsea position estimate. We will describe our recent developments in using these Doppler measurements to aid glider navigation, comparing the results to previous developments in range-aided navigation.

2:05

2pUWa4. Wavenumber domain acceleration for stripmap fast factorized backprojection beamforming. Timothy Marston and Daniel Plotnick (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

Synthetic aperture sonar (SAS) imaging is frequently used in conjunction with autonomous underwater vehicles (AUV's) to perform high resolution wide-area surveys. Wave action and currents perturb the trajectories of AUV's performing these surveys and these perturbations have important ramifications for the SAS beamforming process. Generalized beamforming algorithms operating in the time domain are very flexible with regard to trajectory non-linearities, but they also tend to have a higher computational cost than alternative frequency domain algorithms. Of the generalized beamformers one of the more efficient is fast factorized backprojection (FFBP), which leverages sub-aperture processing and a series of coordinate transformations to improve computational efficiency. FFBP can be used to beamform SAS data, however many SAS systems make use of a real aperture to boost range potential. The existence of a real aperture implies that a frequency domain beamformer could be used to bypass the first few stages of FFBP without loss of generality; however, standard frequency domain SAS beamforming algorithms are not compatible with the FFBP framework. A wavenumber domain beamformer compatible with the early-stage FFBP framework is derived and beamforming results for the modified FFBP algorithm are compared with the standard approach that operates entirely in the time domain.

2:25

2pUWa5. Monitoring of macroalgae (kelp) farms with autonomous underwater vehicle-based split-beam sonar. Erin Fischell, Timothy K. Stanton, Amy Kukulya, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., WHOI, MS 11, Woods Hole, MA 02543, efischell@whoi.edu)

The long-term goal of ARPA-E's MARINER (Macroalgae Research Inspiring Novel Energy Resources) program is to increase the scale of offshore kelp aquaculture so that biofuel may be derived from macroalgae. An autonomous underwater vehicle (AUV) system including acoustic, optic, and environmental sensors has been developed for the purposes of monitoring these large-scale kelp farms. The primary sensor for observing farm infrastructure such as horizontal longlines (from which the kelp grows) as well as kelp growth is a broadband split-beam sonar system. Structural information, such as droop in the longlines, is available from time-of-flight of the acoustic echoes. Growth of kelp is quantified from both time-of-flight and volume backscattering echo data. Experimental results from ocean measurements taken with the REMUS 100 AUV on longlines are presented, along with preliminary processing techniques for estimating longline position and macroalgae extent for vehicle control and farm mapping purposes [work supported by ARPA-E.]

2:45–3:00 Break

3:00

2pUWa6. A hybrid time and wavenumber domain algorithm for generalized beamforming with circular synthetic aperture data. Timothy Marston and Daniel Plotnick (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

Synthetic aperture sonar systems typically scan along a linear trajectory, however autonomous underwater vehicles (AUV's) provide a useful platform for performing scans along more complicated trajectories with specific advantages. One example is the circular scan, which provides superior wavenumber coverage and resolution for images reconstructed near the circle center. A variety of approaches have been put forward for beamforming synthetic aperture data collected along a circular trajectory, but for images that span a significant fraction of the circle radius, accelerated time-domain approaches such as fast-factorized backprojection (FFBP) have been suggested. The computational requirements for generating a fully sampled image collected around a circular aperture are extremely large, however, even for FFBP. This is primarily because of the extremely small pixel size required to maintain adequate spatial sampling as larger portions of the aperture are coherently fused in the later stages of the FFBP algorithm. A method is presented for replacing the later stages of the algorithm with a series of spectral mappings, retaining the generality of the beamformer while improving computational efficiency.

Contributed Papers

3:20

2pUWa7. Underwater source localization using a fleet of three gliders. Yong-Min Jiang (Res. Dept., NATO - STO - Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, yongmin.jiang@cmre.nato.int)

The NATO Centre for Maritime Research and Experimentation has been exploring the potentials of passive acoustic payload equipped underwater gliders for maritime intelligence, surveillance and reconnaissance missions. During the CMRE Glider Sensors and Payloads for Tactical Characterization of the Environment 2015 sea trial, it was successfully demonstrated that a fleet of three passive acoustic payload equipped gliders simultaneously detected the 'anomalous' sound in shallow water and reported back to the glider control center. This work studies the feasibility of localizing the source (the coordinates and the depth) using the information provided by the glider fleet. The results of three different source localization approaches are compared. The impact of the errors in conventional glider time and location data, as well as insufficient/incorrect knowledge of the ocean environment on underwater source localization is discussed. (*Work funded by NATO-Allied Command Transformation.*)

3:35

2pUWa8. A method for passive localization of three-dimensional position. Mingyu Song, JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing CuiWei St., Beijing, China, 56608812@qq.com), and Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang, China)

The movement operation of the machine installed on the ship and the ship itself will disturb the surrounding medium, and the vibration's transmission forms the sound field of a ship. Analyzing the sound field of the target source has profound significance. The target three-dimensional position can be known according to the source of the noise radiated sonar-gram, this can be used for positioning the moving target source and tracking its trajectory A method using focused beam-forming to measure the distribution image (under-water image) of ship's noise source which depth is unknown is presented in this paper. This is for the past open literature mentioned using a one-dimensional horizontal array for two-dimensional plane of the target source localization defects. The arrays cloth on the seafloor, when basing on the principles of geometrical acoustic and the superposition of important early reflected sound energy on direct sound energy, the three-dimensional coordinates of the target would be calculated. For the mentioned principle, this paper carried out the related analyzes simulation and the acoustic image verifies the feasibility of the algorithm. The distance would bring in focused-peak broadening, positioning-error increasing, by dividing frequency band, the phenomenon could be restrained and the problem could be improved. The results show that the method can locate the target quickly and accurately, and it can be used in the fixed arrays offshore areas.

3:50

2pUWa9. Passive detection parameter measurement of azimuth and frequency based on single vector sensor. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Mingyu Song, and JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing, China)

This paper mainly studies the measurement technology of passive detection parameters based on single vector sensor. 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. In this paper, the mathematical model of the radiation noise of the ship is simulated and the demodulation of the ship's radiated noise is carried out by using the square demodulation and the absolute value demodulation. 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 verification of 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.

4:05

2pUWa10. Mapping underwater noise, detection of ships and cetaceans using a *SeaExplorer* glider at a basin level: Feedback from the first 1000 km-long acoustic exploration of the Western French Mediterranean Sea. Laurent Beguery (ALSEAMAR, Rousset, France), Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Romain Tricarico (ALSEAMAR, Rousset, France), Cédric Gervaise, Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France), cathy anna valentini poirier, and Pierre Boissery (Agence de l'Eau RMC, Marseille, France)

In response to concerns about the impact of man-made noise on marine ecosystems, researchers and environmental managers are currently collecting *in situ* measurements of oceanic noise levels. The objectives of *in situ* measurements are to provide the acoustic signatures of individual ships, with the use of AIS databases, to feed the models; to calibrate the model for mapping of shipping noise, and to assess marine biodiversity through the sounds emitted by marine animals (invertebrates, fishes, and cetaceans). The usefulness of the data collected depends on the duration of acquisition and the diversity of the measurements (e.g., shipping density and water depth). Gliders are ideal vehicles to collect noise level data across oceanic basins and over long time periods. Here, we show results from a *SeaExplorer* glider equipped with a high quality acoustic payload travelling for 30 days (09/15/2017-10/15/2017) along a 1000 km-long transect of the Western French Mediterranean Sea. The trajectory of the glider was chosen to sample the highest and lowest shipping densities. We here report on the statistical distribution of oceanic noise levels in the bandwidths assessed by the European Marine Framework Strategy Directive as well as on the detection of cetacean and ship sounds along the transect.

Session 2pUWb

Underwater Acoustics: Signals and Systems II

Anthony P. Lyons, Chair
Center for Coastal and Ocean Mapping, University of New Hampshire, Durham, NH 03824

Contributed Papers

1:30

2pUWb1. Effects of reverberation on estimates of synthetic aperture sonar multi-look coherence. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu), Jonathan L. King (Naval Surface Warfare Ctr., Panama City, FL), and Daniel C. Brown (Appl. Res. Lab., Penn State Univ., State College, PA)

Multi-look coherence makes use of phase information for target detection and classification by splitting the angle and frequency spectral bandwidth contained in a complex synthetic aperture sonar image into sub-bands and then estimating the coherence between the images formed from these sub-bands. Based on experimental results, it appears to be possible to separate man-made targets from interfering reverberation and clutter using coherence, as targets have features that scatter coherently as a function of angle or frequency. Characteristics of a target object may also be inferred, as sub-band coherence is a sensitive function of both angle and frequency. The expectation operation performed to obtain estimates of the complex correlation coefficient is evaluated using a spatial average, which lowers the spatial resolution of the resulting coherence map. The final resolution of a coherence image will be a function of the original image resolution, the number of sub-look images formed from the original image and the number of samples used in the expectation. In this work, we explore the effects of signal, reverberation, and noise levels on estimates of coherence using experimental data. Trade-offs in terms of signal-to-reverberation, expectation window size, and the number of sub-looks will be presented.

1:45

2pUWb2. Compressive normal mode estimation in shallow water. Yongsung Park, Woojae Seong (Seoul National Univ., Gwanak-gu, Gwanak-ro 1, Seoul National University Bldg. 36 - Rm. 212, Seoul 08826, South Korea, ysparkwin@snu.ac.kr), and Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Estimation of normal modes in an ocean waveguide from acoustic data on a vertical line array involves estimation of mode shapes and their wavenumbers. Since these horizontal mode wavenumbers are sparse, compressive sensing (CS) can solve the sparse signal reconstruction problem with high accuracy. Here, grid-free CS technique, atomic norm minimization (ANM) is used to estimate the horizontal mode wavenumbers and their mode shapes. These mode shapes and wavenumbers can be estimated using the grid-free CS without *a priori* knowledge of the environment. The grid-free CS successfully extracts mode shapes and wavenumbers even with a partial water column spanning array and even in the case when the mode shape functions are not orthogonal.

2:00

2pUWb3. Analysis and interpretation of underwater acoustic data originated from the vicinity of the last known location of the Argentinian submarine ARA San Juan. Peter L. Nielsen, Mario Zampolli, Ronan Le Bras, Pierrick Mialle, Paulina Bittner, Alexander Poplavskiy, Mikhail Rozhkov, Georgios Haralabus, Elena Tomuta, Randy Bell, and Patrick Grenard (Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Office E0565, VIC, P.O. Box 1200, Vienna 1400, Austria, Peter.Nielsen@ctbto.org)

The CTBT International Monitoring System (IMS) is a world-wide network of seismic, infrasound, hydroacoustic, and radionuclide stations designed and deployed to detect nuclear explosions. Two of the IMS hydrophone stations, one in the Atlantic Ocean and one in the southern Indian Ocean, recorded signals of unknown nature which originated from the vicinity of the last known location of the San Juan submarine on 15 November 2017. To verify the accuracy of this hydroacoustic event localization, the Argentinian Navy successively deployed a controlled depth charge to the North of the last known position of the submarine. The signals from this source were also detected on the same two IMS hydrophone stations. Several techniques were employed to compare hydroacoustic features in the signals from the 15 November event and the controlled depth charge, including an assessment of spectral energy levels, cepstral analysis, azimuth and arrival time estimation of direct and reflected signals. This presentation compares the received signals, examines their main characteristics, and, through the use of acoustic modelling, identifies possible propagation paths between the source and receiver, including a plausible explanation of early arrivals hypothesized to follow propagation paths through the Antarctic ice sheet.

2:15

2pUWb4. Studying underwater sound level caused by bridge traffic in Lake Washington. Shima Abadi, Derek Flett, Ryan Berge, Jeremy DeHaan, Virdie Guy, Urooj Qureshi, and Michael Cook (Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, flettd23@uw.edu)

Underwater noise pollution due to human activities has greatly increased in recent years. There are several studies on high-intensity impulsive noises such as pile driving and seismic exploration. However, less is known about the effects of long-term exposure to low-intensity noises such as those due to bridge traffic. To characterize such noises, we have designed an underwater recording package. This package is equipped with a hydrophone to measure noise, a Raspberry-Pi to control recordings, and a cellular modem to stream near real-time data to the cloud. Data is collected in Lake Washington at two locations: (1) close to the Evergreen Point floating bridge across Lake Washington that connects Seattle to its eastern suburbs and (2) far from the bridge in the middle of the lake for comparison. The Evergreen Point floating bridge, the longest floating bridge in the world, is made of 77 large water-tight concrete pontoons and capable of carrying over 70,000 vehicles per day. The noise level measured close to this bridge is correlated with the public data on traffic load and wind speed available by the Washington State Department of Transportation.

2:30

2pUWb5. Transverse horizontal spatial coherence of explosive signals in northern South China sea. Bo Zhang, Jingyan Wang, and Yanjun Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhangbo@mail.ioa.ac.cn)

Transverse horizontal spatial cross-coherence was measured experimentally by bottom-mounted receiver arrays using explosive sources in Northern South China sea, where the water depth varies from 100 m to 1000 m in about 100 km ranges with a 200 m depth at about 75 km range. Expressed in terms of wavelengths, the coherence length is shown to be larger than 150λ at an acoustic frequency of 100 Hz in shallow water of 100 m depth. It is much greater than Carey's shallow-water experimental result 30λ estimated from array signal gain (*The Journal of the Acoustical Society of America 104, 831 (1998)*), while it is consistent with Rouseff's modeling result of a coherence length larger than 200λ (*The Journal of the Acoustical Society of America 138, 2256 (2015)*). In contrast to the good coherence in shallow water, the coherence length is measured to be only 5λ ~ 20λ at 100Hz in the transitional area with the array bottom-mounted at 1000m depth. Besides, both Carey and Rouseff argue that the transverse horizontal spatial coherence length depends only weakly on range. In the present study, however, the coherence length is shown to depend highly on range, and it fluctuates synchronously with the sound-field intensity while range varies.

2:45–3:00 Break

3:00

2pUWb6. Empirical modelling for derived metrics as function of sound exposure level in marine pile driving. Roel A. Müller (Acoust. & Sonar, TNO, Oude Waalsdorperweg 63, Den Haag 2597 AK, Netherlands, roel.muller@tno.nl), Michael A. Ainslie (JASCO Appl. Sci. (Deutschland) GmbH, Eschborn, Germany), Michele B. Halvorsen (CSA Ocean Sci. Inc., Stuart, FL), and Tristan Lippert (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Hamburg, Germany)

To protect marine life, piling activities at sea are subject to regulations on sound emission. Different regulatory authorities base different action criteria based on different statistics of the emitted sound pressure. Modelling efforts to predict these metrics have achieved mixed success. While the sound exposure level (SEL) can be predicted relatively well, it has proven harder to model the peak sound pressure level (Lpk) accurately. It would therefore be valuable to have a reliable way of estimating Lpk, based on a prediction of SEL. Correlations between SEL and Lpk and between SEL and rms sound pressure level (Lrms) are obtained from measurements during piling of the Luchterduinen wind farm off the Dutch west coast, and are used to assess the applicability of correlations found at other wind farms. A metastudy using data from Luchterduinen as well as the German Bight gives a more robust trend line for use at other North Sea sites. Lrms is computed in two different ways, involving the 90 % energy signal duration on the one hand and the effective signal time duration (Teff) on the other. The value of Lrms based on Teff is found to be more robust. [Work sponsored by BOEM.]

3:15

2pUWb7. *In situ* demonstrator of a method to track in three dimensions cetaceans with passive acoustics in the context of close interactions impacts with marine renewable energy devices. Julie Lossent (Inst. of Eng. Univ. Grenoble Alpes, CNRS, Grenoble INP, GIPSA-Lab,, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Bouzidi Medjber (SINAY, Caen, France), Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Achraf Drira, Yanis Souami (SINAY, Caen, France), and Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France)

Collision risk and close interactions of cetacean with marine renewable energy (MRE) devices are among the main concerns and the least documented of the list of potential impacts of MRE. In this study, we developed a method to track in three dimensions click trains emitted by cetaceans in a close vicinity to MRE devices. We deployed two synchronized directional recording arrays 150 m apart from each other near Cherbourg (France). With an acoustic tag VEMCO© placed at several depths we performed transects with several trajectories simulating the potential behaviors (diving, surfacing, avoidance, and attraction) and click production of *delphinidae*. Thanks to the directional recording arrays and by crossing the direction of arrival of the same sounds calculated on each array, we build 3D maps of the synthetic clicks. We manage to reconstruct the trajectories (absolute positions x, y, and z in meters) of the emitter. In the areas located 100 m upstream and downstream of the two recording arrays, the location accuracy is maximum. This method could also be used to prove the departure and the non-return of cetacean from an exclusion zone. Moreover, this approach has been designed to be operational for the monitoring of cetaceans in environmental impact assessment studies.

3:30

2pUWb8. Controlled source level measurements of whale watch boats and other small vessels. Jennifer L. Wladichuk, David E. Hannay, Zizheng Li, Alexander O. MacGillivray (JASCO Appl. Sci., 2305 – 4464 Markham St., Victoria, BC V8Z 7X8, Canada, jennifer.wladichuk@jasco.com), and Sheila Thornton (Sci. Branch, Fisheries and Oceans Canada, Vancouver, BC, Canada)

The Vancouver Fraser Port Authority's Enhancing Cetacean Habitat and Observation (ECHO) program sponsored deployment of two autonomous marine acoustic recorders (AMAR) in Haro Strait (BC), from July to October 2017, to measure sound levels produced by commercial vessels transiting the strait. Fisheries and Oceans Canada (DFO), a partner in ECHO, supported an additional study using these same recorders to systematically measure underwater noise emissions (0.01-64 kHz) of whale watch boats and other small vessels that operate near Southern Resident Killer Whale (SRKW) summer feeding habitat. During this period, 20 different small vessels were measured operating at a range of speeds (nominally 5 knots, 9 knots, and cruising speed). The measured vessels were categorized into six different types based primarily on hull shape: ridged-hull inflatable boats (RHIBs), monohulls, catamarans, sail boats, landing craft, and one small boat (9.9 horsepower outboard). Acoustic data were analyzed using JASCO's PortListen® software system, which automatically calculates source levels from calibrated hydrophone data and vessel position logs, according to the ANSI S12.64-2009 standard for ship noise measurements. We found clear positive correlations of source levels with speed for all of these vessels; however, the speed trends were not as strong as those of large commercial vessels.

3:45

2pUWb9. Characterization and modeling of typhoon-generated noise in South China Sea. Jingyan Wang and Fenghua Li (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, 100190 Beijing, China, wangjingyan@mail.ioa.ac.cn)

Ocean noise recorded during a typhoon is dominated by wind-generated noise sources, and can be used to monitor the typhoon and investigate the mechanism of the wind-generated noise. An analytical expression for the typhoon-generated noise intensity is derived as a function of wind speed. A "bi-peak" structure was observed in an experiment in South China Sea during which typhoon-generated noise was recorded. Strong wind speed dependence and frequency dependence were also observed in the frequency range of hundreds to thousands of Hertz. The model/data comparison shows that results of the present model are in reasonable agreement with the experimental data, and the typhoon-generated noise intensity has a dependence on frequency and a power-law dependence on wind speed.

Session 2pUWc

Underwater Acoustics: Effects Due to Elasticity and Interfaces

Ahmad T. Abawi, Chair
HLS Research, 3366 North Torrey Pines Court, Suite 310, La Jolla, CA 92037

Contributed Papers

2:30

2pUWc1. Dynamic permeability and tortuosity in weakly consolidated granular media. Peyman M. Moradi and Apostolos Kantzas (Chemical and Petroleum Eng. Dept., Univ. of Calgary, ES903-2500 University Dr. NW, Calgary, AB T2N1N4, Canada, seyedpeyman.mohammad@ucalgary.ca)

Micro-scale rock heterogeneity plays an important role in modifying the seismic response of porous media and must be considered in reservoir studies such as seismic monitoring of fluid flow or interpretation of Stoneley waves. This work focuses on the prediction of dynamic permeability and dynamic tortuosity of granular media at different frequencies from porosity and average grain size data, utilizing numerical simulations. The simulations are conducted through cubic consolidated and unconsolidated digital rock samples of three different rock types, by applying a macroscopic oscillatory pressure gradient to each porous medium and combining fluid flow with solid mechanics to solve the wave propagation problem. The JKD parameter called dynamically connected pore size (Λ) is then predicted for each rock type. We investigate how pore space geometry, cement content and morphology, and matrix heterogeneity can influence the dynamic and geometrical tortuosities.

2:45

2pUWc2. Scattering from multiple elastic targets using the coupled finite element/boundary element method. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com), Ivars P. Kirsteins (NUWC, Newport, RI), Philip L. Marston, and Timothy D. Daniel (Washington State Univ., Pullman, WA)

The fluid-structure interaction technique provides a paradigm for solving scattering from elastic targets embedded in a fluid by a combination of finite and boundary element methods. In this technique, the finite element method is used to compute the target's elastic response and the boundary element method with the appropriate Green's function is used to compute the field in the exterior medium. The two methods are coupled at the surface of the target by imposing the continuity of pressure and normal displacement. This results in a self-consistent boundary element method that can be used to compute the scattered field anywhere in the surrounding environment. The method reduces a finite element problem to a boundary element one with drastic reduction in the number of unknowns, which translates to a significant reduction in numerical cost. In this talk, the method is extended to compute scattering from multiple targets by self-consistently accounting for all interactions between them. The model allows to identify block matrices responsible for the interaction between targets, which proves useful in many applications. The model is tested by comparing its results with those measured involving two aluminum cylinders one of which is excited by modulated radiation pressure.

3:00

2pUWc3. Seismic profile processing to identify sediment layering structure in shallow water. Wenbo Wang, Qunyan Ren, Jun Li, and Li Ma (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renqunyan@mail.ioa.ac.cn)

The equivalent layering structure of surface sediment is a key parameter for sound propagation modelling and geoacoustic inversion in shallow waters. Previously, the structure identification is manually performed through identifying these horizontal lines as shown in the profiling image, which represent the interfaces with strong acoustical impedance contrast. In this paper, a hybrid image processing method (statistical equalization, multi-scale line filtering, and wavelet decomposition) is adapted to automatically identify these interfaces, afterwards, an image-binary threshold and a dislocation phase subtraction approach are successively applied to determine sediment layer number and relative thickness. The data from dozens lines of surveys as collected in 2016 are processed. Through integrating the processing results with GPS data, 2D distributions of sediment layer number and thickness of the first and second sediment layer are constructed. The spatial distributions of sediment layer number and thickness demonstrate certain trends, which are in accordance with expected sedimentary process in this region and should be carefully treated in 3D sound propagation modelling and geoacoustic inversion.

3:15

2pUWc4. Reflected seismic waves radiated by the apex of a wedge-shaped ocean. Wang Xiaohan, Shenchun Piao, Ya-Qin Liu, and Dong Yang (Haerbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, wangxiaohan@hrbeu.edu.cn)

A sound propagation experiment has been carried out near the coast in Qingdao to investigate underwater sound propagation characteristics over a sloping bottom. Hydrophones and Ocean Bottom Seismometers (OBSs) were arranged in different place to receive the signal radiated by explosive source. In the temporal signals received by the OBSs, some weak and mystical "tails" following the direction waves were observed but the hydrophones only received the direction waves without the "tails." Analyzing signal of different OBSs at different distance, it is indicated that the "tails" is the seismic waves coming from a fixed location. The temporal signal propagating in elastic wedge-shaped ocean is simulated with spectral-element method. The results of the model show that, when the acoustic waves propagates to the apex of the wedge, the boundary among the surface, the seabed and the coast, the apex will radiate the "tail" seismic waves. The reflection seismic waves can be apply in geoacoustic parameter estimation and Underwater Topographic Survey.

3:30

2pUWc5. An examination of the scaling rules for sound emissions from targets excited by point-like forces. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, i.kirsteins@gmail.com), Ahmad T. Abawi (HLS Inc., La Jolla, CA), Philip L. Marston, and Timothy D. Daniel (Washington State Univ., Pullman, WA)

In earlier work [T. Daniel *et al*., JASA **140**, 3123 (2016)] we had showed experimentally that the modes of small elastic targets in water could be excited using modulated radiation pressure (MRP) generated by focused ultrasound to create detectable sound emissions. A potential advantage of the MRP approach is that the narrow beam of the ultrasound beam permits the generation of point-like forces on the target's surface with surgical precision. But an important question is how this technique scales with target size, e.g., the amount of force required to achieve a desired sound emission level. Here we examine how the sound emission levels and surface velocities of target modes driven by a point force scale with target size a. We will show that for a constant driving force, the far-field sound emission pressure levels scale as 1/a and the surface velocity as $1/a^2$. Specific examples are presented for circular plates, spheres, and cylindrical-like targets. Finally, results will be shown from a recent tank experiment at Washington State University where we attempted to experimentally determine the scaling rules using small targets excited by MRP.

3:45–4:00 Break

4:00

2pUWc6. The study on dispersion characteristics of Scholte wave in elastic media with variable parameters. Ya-Qin Liu, Hai-Gang Zhang, and Shi-E Yang (Harbin Eng. Univ., No.145 Nantong St., Nangang District, Heilongjiang, Harbin 150001, China, liuyaqin@hrbeu.edu.cn)

In the real ocean environment, due to the reflection of the water-sediment interface, the elastic sediment will generate inhomogeneous compressional wave and inhomogeneous shear wave, and the interaction between them will form the Scholte wave. Besides, the compressional and shear wave velocities in an elastic sediment layer are changing with depth leading to the coupling of compressional and shear waves. As the dispersion characteristics of the Scholte wave is affected by the coupling effect, in this paper, a study on the dispersion characteristics of Scholte wave in an environment with variable parameters elastic sediment layer is given. Specific environmental examples are simulated and the dispersion curves are compared with the KRAKENC program. In addition, the influence of the parameters of elastic sediment layer on the Scholte wave dispersion curves is analyzed. Comparing the dispersion curve of the Scholte wave in the uniform sediment layer and in elastic medium with variable parameters, it is found that some characteristics of the dispersion curve in the uniform layer will no longer fit the elastic layer with variable parameters.

4:15

2pUWc7. Broadband numerical simulation of scattering from rough pressure-release and fluid-fluid interfaces. Derek R. Olson (Oceanogr., Naval Postgrad. School, 201 Appl. Sci. Bldg., University Park, Pennsylvania 16802, olson.derek.r@gmail.com) and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

The scattered field from the seafloor is often measured using short, broadband pulses, whereas many models for the mean scattered intensity are in the frequency domain. This As higher resolution seafloor mapping systems, i.e., synthetic aperture sonar, become more common, it is important to understand the effects of high resolution on measurements of both the averaged scattered intensity, and the distribution of the envelope of the scattered field. To address this problem, Monte Carlo simulations of the scattered field due to rough interfaces separating two fluids are perform. The integral equation governing the total pressure, the Helmholtz-Kirchhoff integral equation, is numerically solved using the boundary element method. Simulations are performed both in the limit of pressure release, as well as for more realistic sediment sound speed and density parameters. The von-Karman spectrum is used to generate random rough surfaces. Fourier synthesis is used to to generate time domain signals, which are then analyized in terms of the mean energy and amplitude distribution. The dependence of the scattering cross section and scintillation index on signal bandwidth (resolution) is examined. It is found that there is no observable dependence of the scattering cross section on bandwidth, but that there is significant dependence of the scintillation index.

4:30

2pUWc8. Multiple scattering effects for a solid cylinder with axis oblique relative to a nearby horizontal surface. Aubrey L. Espana, Daniel Plotnick, Kevin Williams (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

It has been shown that when studying the acoustic scattering from a cylinder near a flat interface, there exists multiple paths by which sound can travel to, and subsequently scatter from, the cylinder. Specifically, sound can couple into a number of surface elastic waves (SEW), some of which give rise to an observed enhancement to the backscattering. Previously, the coupling conditions for these mechanisms have been derived, which are dependent on the relative angles between the target, the source/receiver and the interface [Plotnick, *et al*., JASA 140(3), 2016]. Here, a non-singular Helmholtz Kirchhoff Integral method has been implemented in a hybrid finite element/propagation model in order to predict the first and second order scattering mechanisms that exist when the cylinder is near an air-water interface, with its axis oblique with respect to the interface. This model, combined with the coupling conditions for SEWs, is used to dissect tank measurements that previously were only partially understood. Various multiple scattering mechanisms are identified and discussed. [Work supported by the Office of Naval Research.]

4:45

2pUWc9. Analysis on acoustic longitudinal horizontal correlation of air-to-water transmission. Lingshan Zhang, ZhaoHui Peng, and GuangXu Wang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, zhanglingshan@mail.ioa.ac.cn)

Spatial correlation is an important characteristic of underwater sound field, which is relevant to the detection of aerial targets from underwater. But compared with waterborne sources there are few researches about airborne sources. In order to perform further analysis on the air-to-water sound transmission in shallow water, State Key Laboratory of Acoustics, Institute of Acoustics, conducted an experiment in the South China Sea in March, 2013. During the experiment, multi-frequency signals transmitted by a hooter hung on a research ship were received by a horizontal line array (HLA) which laid on the seabottom, and signals far from 9.8 km were received successfully. Based on data recorded in the experiment, the longitudinal horizontal correlation coefficients of underwater sound field excited by airborne source at different distances with the frequency is 128 Hz are obtained, which have evident oscillation structure as horizontal spacing increases. The mathematical expression is derived based on the normal mode theory and gives a better physical explanation for the oscillatory behavior of the experimental longitudinal horizontal correlation.

5:00

2pUWc10. The treatment of the density discontinuity in the split-step Fourier algorithm using the hybrid approach. Mustafa Aslan (Turkish Naval Acad., National Defense Univ., 833 Dyer Rd., Monterey, CA 93943-5216, maslan1@nps.edu)

Various numerical models generate the solution for the modelling of underwater acoustic wave propagation treat the density discontinuity due to the density change at the interface by different approaches. In the split-step Fourier parabolic equation model, the smoothing function can be applied for the treatment of the density discontinuity, but the phase error is also accumulated with range in the solution. Yevick and Thomson developed the hybrid split-step/finite difference method in which density dependent terms are introduced in the additional operator to be applied by Padé approximation in the algorithm and the phase accuracy is significantly improved using the method. In this work, instead of Padé approximation, various approximations is examined for the density related operator in the split-step Fourier algorithm.

5:15

2pUWc11. Three dimensional finite element modeling of scattering by objects at rough interfaces. Aaron M. Gunderson and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78758, aaron.gunderson01@gmail.com)

Fully three-dimensional finite element models for buried objects and objects at interfaces have been extended to incorporate interface roughness. Near field results are solved using a fully scattered field formulation, in which all fields in the absence of the target are applied and the target scattering is solved. Results are extended to the far field through the Helmholtz-Kirchhoff integral, using a numerically determined Green's function approach. Interface roughness is applied using a modified power law wavenumber spectrum. Previous model demonstrations evaluated the scattering in the flat interface limit, in which background fields were prescribed analytically. Now, the presence of the surface roughness forces background fields to be evaluated and prescribed numerically. Such models have strong applicability for predicting scattering by objects buried within or resting on a rough seafloor interface. Model results are compared with various experimental and other modeled results, demonstrating model validity over a wide frequency range. [Work supported by Applied Research Laboratories IR&D and ONR, Ocean Acoustics.]

OPEN MEETINGS OF TECHNICAL COMMITTEES/SPECIALTY GROUPS

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, 6 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Rattenbury A/B (FE)
Acoustical Oceanography	7:30 p.m.	Esquimalt (VCC)
Animal Bioacoustics	7:30 p.m.	Oak Bay 1/2 (VCC)
Architectural Acoustics	7:30 p.m.	Theater (VCC)
Physical Acoustics	7:30 p.m.	Colwood 1/2 (VCC)
Psychological and Physiological Acoustics	7:30 p.m.	Salon B (VCC)
Speech Communication	7:30 p.m.	Salon A (VCC)
Structural Acoustics and Vibration	7:30 p.m.	Saanich 1/2 (VCC)

Committees meeting on Wednesday, 7 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	(FE)
Signal Processing in Acoustics	7:30 p.m.	(FE)

Committees meeting on Thursday, 8 November

Committee	Start Time	Room
Computational Acoustics	4:30 p.m.	Esquimalt (VCC)
Musical Acoustics	7:30 p.m.	Crystal Ballroom (FE)
Noise	7:30 p.m.	Shaughnessy (FE)
Underwater Acoustics	7:30 p.m.	Rattenbury A/B (FE)

Session 3aAA

Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration: Advances in the Laboratory Testing of Materials

Ronald Sauro, Chair
NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541

Chair's Introduction—7:55

Invited Papers

8:00

3aAA1. Common problems associated with reverberation rooms in sound testing laboratories. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Reverberation rooms have proven to be a very useful tool in acoustic laboratories for measuring the sound absorption of acoustic materials, airborne and impact sound transmission through building partitions, as well as measuring the overall radiated sound power level of a wide variety of sound sources. The primary benefit of a reverberation room is that the measurement can be made in much less time than with alternative environments (e.g., anechoic or hemi-anechoic chambers, or via acoustic intensity). The primary disadvantage of the reverberation room measurements is that all information regarding the directivity of the acoustic source is lost. This paper discusses some of the design and acoustical performance problems that are often found in reverberation rooms that could have been avoided with proper foresight. Issues that will be discussed include location relative to nearby noise and ground vibration sources, room size and shape, space planning for microphone locations, low frequency absorbers, and the importance of temperature and humidity control. Specific guidelines that should be followed when planning a new reverberation room will be provided.

8:20

3aAA2. Measuring with noise? We can do better!. Markus Müller-Trapet and Christoph Hoeller (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, Markus.Mueller-Trapet@nrc-cnrc.gc.ca)

It has long been established that correlation-based measurement techniques, using, e.g., maximum length sequences and swept sines, outperform classical methods based on random Gaussian noise. Among the advantages of these modern techniques are better rejection of background noise and reduced measurement duration to achieve similar or better precision. These advantages are especially interesting in room and building acoustics, where many measurement positions typically have to be covered and background noise is often an issue, especially when measuring sound transmission loss or velocity level differences. Unfortunately, there is currently no provision in the ASTM standards on room and building acoustics that would allow the use of these modern measurement methods. To demonstrate the advantages of these methods, this contribution will present an example of measurements of the apparent sound transmission loss, i.e., measurements of sound pressure level differences and reverberation times. In addition to the standardized measurements according to ASTM E336, all measurements were repeated with maximum length sequences and swept sines. The results will be compared and the advantages of the modern techniques will be highlighted.

8:40

3aAA3. Designing reverberation chambers for improved low frequency response. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Most reverberation rooms are able to provide good diffusion and spatial uniformity in the reverberant field of the source at mid- and high-frequencies. Meeting the broadband spatial uniformity requirements for ANSI S12.51 (ISO 3741) is relatively easy to achieve in the frequency bands 100 Hz to 10,000 Hz provided that the recommended room volumes are used. However, meeting the spatial uniformity requirements becomes much more difficult when the desired frequency range includes frequency bands below 100 Hz and the room must meet the pure tone qualification criteria (as is required in AHRI 220). There are two techniques that can be used to improve the low frequency room response of a reverberation room. One is to select room dimensions that will provide near uniform spacing of the room modes in the low frequency bands. The second technique is to install low frequency absorbers to the surfaces of the reverberation room to lower the "Q" of the individual modes. This paper will re-introduce a new quantity (first introduced at Internoise 2012 in New York City) called the modal distribution factor, D^2, and illustrate how it can be used to optimize the dimensions of any reverberation room within the space constraints of the proposed location. In addition, the paper will illustrate the design and acoustical performance of two very successful reverberation rooms that were conceived and constructed using these techniques.

9:00

3aAA4. Design of a new test chamber to measure the absorption, diffusion, and scattering coefficients. Peter D'Antonio (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), Cheol Ho Jeong, and Mélanie Nolan (Danish Tech. Univ., Kgs. Lyngby, Denmark)

This presentation will describe the design for a new 384 m^3 test chamber that will allow the measurement of the random incidence absorption coefficient, according to ISO 354, the scattering coefficient, according to ISO 17497-1, the diffusion coefficient, according to ISO 17497-2 and the normal incidence low frequency absorption coefficient, according to ISO 10534-2. The rev room design incorporates optimal modal dimensional ratios in accordance with (T. Cox, P. D'Antonio & Avis, J. Audio Eng. Soc. Vol. 52, No. 6 (June 2004)), both boundary and hanging panel diffusers to allow satisfactory diffusivity, in collaboration with the research at DTU. Diffusivity will be verified using a reference absorber, consisting of 100 mm porous absorber with a 100 mm cavity. A sufficiently large reflection free volume is provided in the chamber for diffusion coefficient scale model measurements. The diffusion goniometer and 12 m^2 absorber samples are rotated in accordance with test. Scattering coefficient measurements are made according the ISO 17497-2 and the correlation method, using correlated polar responses. A steel and concrete lined low frequency impedance tube measuring 600 mm x 600 mm x 5.8 m long and 7 tons will be used to measure absorption from 25 to 250 Hz.

9:20

3aAA5. How to create an advanced acoustical test and research laboratory in 5 months. Ronald Sauro (NWAA Labs, Inc, 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com) and Jerry G. Lilly (JGL Acoust., Inc., Issaquah, WA)

This paper will take you through the need, design, building, and testing of a state of the art acoustical testing and research facility in a 5 month period. We will discuss the increasing need for wide frequency bandwidth measurements of absorption, transmission loss, very low level sound power measurements, and high resolution measurement of the directivity of speakers and diffusers. We will also look at the limits of standard laboratories and observe what changes are needed to extend those abilities and limits. At the end of the paper, we can look at the end result measurements of these facilities to see how close to design and performance the lab approached.

9:40

3aAA6. An investigation of laboratory sound transmission loss testing for steel-framed partitions. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Laboratory sound transmission loss (STL) test reports and research publications may be used to validate partition designs as valid and code-compliant sound isolation treatments in buildings. However, longitudinal and cross-laboratory analyses of the sound transmission loss of steel-framed building partitions have revealed such a wide variation in experimental results and calculated ratings that design validation is impossible. There are multiple possible explanations for this variation in results. Steel manufacturers list a variety of mil-thicknesses under the same "gauge" specification, for example. This research study investigates laboratory testing parameters and material components such as the repeated use of the same steel framing for varying panel configurations, screw length and spacing, and steel mil-thickness, regarding the effect that each of these aspects of sample assembly design and testing have on the sound transmission loss of steel-framed partitions.

10:00–10:15 Break

Contributed Paper

10:15

3aAA7. Acoustic characteristics comparison for glasses and windows by laboratory measurement. Hui Li, Jianghua Wang (Beijing Deshangjingjie Technol. Development Ltd. Co., Main Bldg., Rm. 104, Beijing, Beijing 100084, China, lihuisylvia@aliyun.com), Xiang Yan (Architecture, TsinghuaUniv., Beijing, China), and Jing Guo (Beijing Deshangjingjie Technol. Development Ltd. Co., Beijing, China)

Windows are usually with a much lower sound insulation index comparing to walls. Integrated sound insulation of building components is decided by the weakest element. So, windows are the key of indoor quietness. The Rw of 3 structures of glass and 5 structures of windows are measured in laboratory. By comparing the results, the influence of thickness of glass, thickness of air layer, thickness and material of lamination and frame are shown. Rw + Ctr of the glasses are between 34 dB and 39 dB. Rw + Ctr of the windows are between 27 dB and 41 dB.

Invited Paper

10:30

3aAA8. New ASTM ratings for quantifying low and high-frequency impact insulation in laboratory testing. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

The authors have recently published [*J. Acoust. Soc. Am.* **141**, pp. 428–-440 (2017)] a two-rating method for evaluating impact sound insulation, in which the low and high-frequency components of the floor are evaluated independently. Although originally described for field testing, the ratings can be applied to laboratory testing. ASTM draft classifications have been submitted for low and high-frequency versions of Impact Insulation Class, named LIIC and HIIC, respectively. The scope and use of the new ratings are explained.

Contributed Paper

10:50

3aAA9. Look—Do you see the noise leaking through that ceiling? Gary Madaras (Acoust., Rockfon, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com)

Oftentimes, acousticians must convey complex, three-dimensional, acoustical phenomena that occur within and between rooms in buildings to architects, interior designers and other visually oriented people. The message can be lost during translation from quantitative acoustics metrics and their acronyms to the design and intersection of actual building elements such as walls and ceilings. The current phase of the Optimized Acoustics Research Program focuses on turning sound absorption and sound isolation performance visual by using color mapping. Much like a thermal imaging camera shows differences in surface temperatures, a sound intensity probe is being used to produce high definition color sound mapping of noise transmitting through acoustical ceiling systems and sound reflecting off or being absorbed by surfaces with different absorption coefficients. This measurement method helps to bridge the gap between the technical, quantitative side of acoustics and the visual side of design in an impactful and memorable way.

Invited Papers

11:05

3aAA10. Testing diffusion, Why we need to change the way we do things. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

For the last 20 years, different methods have been used in both educational and professional environments to provide test data regarding the performance of diffusers and diffusive products. In many cases, these test methods have defied certain recognized standards common in physics. In other cases, the test methods have been converted to use in test environments which are counter-intuitive to producing the outcomes expected. Additionally, the standard expectation of a single chart producing coefficient results in one or two lines has proven to provide far too little information for the acoustician. This minimal information also does not translate well to actual use in today's computer design programs. New test developments over the last 7 years by the author, in concert with others, have yielded test data that is both informational and multi-dimensional in nature. Providing both magnitude, polar, phase, and other data, this test, being developed under the guidance of ASTM, avoids the complications inherent in the current standards. This presentation will discuss both current standards and new technologies to overcome the limitations of those standards.

11:25

3aAA11. Using laboratory measurement data to improve acoustic simulations and evaluate performance. James J. DeGrandis (Acoust. First, 2247 Tomlyn St., Richmond, VA 23230-3334, jim@acousticsfirst.com), Hassan Azad (Univ. of Florida, Gainesville, FL), and Ronald Sauro (NWAA Labs, Inc., Elma, WA)

There are many different uses for the data that comes from a measurement. With the advent of new testing methodologies, data has become finely granular—allowing for precise analysis of the properties and performance of materials, geometries, and even simulations. Raw measurement data can be retained and compared directly to the output of complex acoustic models for development, improvement, evaluation, and eventually a form of calibration. Advances in technology are rapidly removing the limitations of computational power needed to create accurate models of acoustic phenomena in a timely and efficient manner. This allows a progression from limited particle simulations on single workstations, to large scale wave-based simulations using cloud-based GPU computing clusters. Performance will be evaluated, comparing their output to laboratory measurements, with the goal of creating better tools for acoustic design, prediction, experimentation, and development.

Contributed Paper

11:45

3aAA12. Density in diffusion, quantifying aural quality through testing and listening. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com) and Chips Davis (Chips Davis Designs, Concord, CA)

Diffusion in recording studios and critical listening spaces has been around for well over 30 years. During that time, acousticians and users alike confirm the fact that it makes a difference in the listening space but have a difficult time defining exactly what it is or why they like it. Diffusion test data in the past have been inconclusive as to showing what makes one design more pleasant to the listener than another. This is especially true with *in-situ* testing. The *in-situ* testing shown in the presentation will be concentrated on diffusion density or the amount and quality of diffuse energy being reflected. This presentation will also show a set of experiments between different sets of diffusers used as side diffusion in a recording studio. The experiments, using a binaural head and calibrated microphones, will show both test data as well as recorded responses which will be played back to the listening audience. Lab test data, using the proposed ASTM diffusion test under development, will be included to show the diffuser differences from a laboratory perspective. Attendees will be asked to contribute their perspectives to the recording playbacks to further the experiment as to the listener's experience.

Session 3aAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species I

Thomas F. Norris, Cochair
Bio-Waves Inc., 517 Cornish Dr., Encinitas, CA 92024

Tiago A. Marques, Cochair
School of Mathematics and Statistics, University of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews KY16 9 LZ, United Kingdom

Chair's Introduction—9:00

Invited Papers

9:05

3aAB1. Lessons learned from acoustically tracking baleen whales on the Navy's Pacific Missile Range Facility. Tyler A. Helble, E. E. Henderson (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com), Stephen W. Martin, Gabriela C. Alongi, Cameron R. Martin (National Marine Mammal Foundation, San Diego, CA), and Glenn Ierley (Univ. of California San Diego, Houghton, MI)

Detection, localization, classification, and tracking of marine mammals has been performed on the U.S. Navy's Pacific Missile Range Facility (PMRF) for over a decade. The range hydrophones are time-synchronized, have excellent spatial coverage, and monitor an area of approximately 1200 km^2. Even with these ideal assets, there are hidden challenges when attempting acoustic density estimates of baleen whales on the range. This talk will cover lessons learned from tracking several species of baleen whales on the range over the last decade.

9:30

3aAB2. Using detection received level as a guide for odontocete density estimation. John Hildebrand, Kaitlin E. Frasier, and Alba Solsona Berga (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

Odontocete echolocation clicks have been used for density estimation using cue-counting. We present an approach for assessment of the appropriate threshold for click detection based on the click received sound-pressure-level distribution. Under the assumption of random spatial distribution of echolocating animals with respect to the sensor, the number of detections should decrease with increasing received level. This pattern is verified by simulation modeling, even after incorporating diving behavior and sound production directionality. When measured received level distributions show peaks at values above the detection threshold, errors in the detection process are the most likely explanation. Potential remedies involve setting a higher detection threshold and/or fully characterizing both missed and false positive rates.

9:55

3aAB3. Parameters determining the suitability of bat species for acoustic monitoring. Jens C. Koblitz, Anne Scharf (Max Planck Inst. for Ornithology, Univ. of Constance, Constance 78464, Germany, jkoblitz@orn.mpg.de), Peter Stilz (BioAcoust. Network, Hechingen, Germany), Chloe Malinka (Aarhus Univ., Aarhus, Denmark), and Jamie D. Macaulay (Sea Mammal Res. Unit, Anstruther, Fife, United Kingdom)

Acoustic monitoring of bats is increasingly used in biodiversity assessments, population monitoring, and environmental impact assessments. In addition to accurate species identification, additional factors make it challenging to derive population trends or better sizes based on acoustic monitoring. Inter- and intra-species- as well as individual variation of acoustic parameters and acoustic activity result in varying detection probabilities. Changes in environmental conditions result in large changes in the volume monitored by the device. Differences in the devices used for acoustic monitoring make it inherently difficult to compare data collected with different devices. The single call monitoring volume is modelled for bats belonging to different guilds under consideration of the different call parameters such as call intensity, frequency, and directionality. By broadcasting bat echolocation calls from various distances to monitoring devices, the acoustic parameters influencing the successful detection of a call were examined. A microphone array was used to track bats in the vicinity of monitoring devices and the distance between device and bat was measured for each call based on the time of arrival difference. The acoustic detection function, the probability of detecting calls as a function of distance, was then derived for multiple detector types.

10:20–10:35 Break

10:35

3aAB4. Using passive acoustics to estimate populations of animals in groups. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Passive acoustic monitoring is a widely used method to identify animal species and determine spatial and temporal activity patterns. One area where acoustic methods have not yet been successfully applied, however, is in determining population counts. Recently, my lab developed an acoustic method that accurately predicts the count of bats leaving from a cave roost in a given time period. Data were acquired of Brazilian free-tailed bats (*Tadarida brasiliensis*) across multiple nights and at different cave locations with different roost structures. We used a single microphone to monitor echolocation activity and simultaneously record the emerging bats with thermal and standard video. Bat abundance counts were determined from a single video frame analysis (every 10 sec) and were compared to different energy measures of an acoustic sequence recorded at the time of the analyzed video frame. For most cave locations, linear regression models successfully predicted bat emergence count based on acoustic intensity of the emerging stream, which indicates future population estimates may be collected solely with acoustics. Here, I describe this method and report on its application for counting other animals in groups.

10:50

3aAB5. Estimating population densities of temperate, insectivorous bats based on automatically recorded calls. Markus Milchram and Alexander Bruckner (Integrative Biology and Biodiversity Res., Univ. of Natural Resources and Life Sci., Gregor-Mendel-Strasse 33, Vienna 1180, Austria, markus.milchram@students.boku.ac.at)

Estimating population density based on automatically recorded calls is a key topic in bioacoustics, as individual recognition of animals is impossible. Several recently developed models do not require individual recognition but nevertheless allow to estimate density. However, there is a need to test these models on empirical data. Here, we used generalized random encounter models (gREM) to estimate population densities based on automatically recorded bat calls. To check the validity of the derived estimates, we fit Royle-Nichols models to species detection/non-detection data. Estimates of the two models were compared to each other and to estimates from published studies. The estimates of both models and literature estimates were within the same order of magnitude. Both models give reliable estimates of population density. However, we provide some cautionary notes for practical use: Bats which enter the detection sphere from above might bias results of gREMs, as the model simplifies the detection sphere to a two-dimensional area. On the other hand, reduction to detection/non-detection data in Royle-Nichols models results in information loss, which could limit their applicability in common species. Finally, we recommend to consider species behaviour carefully when applying one of the tested models.

11:05

3aAB6. Estimating density of fin whales and beaked whales with a mix distance sampling and spatially explicit capture recapture design. Tiago A. Marques, Len Thomas (School of Mathematics and Statistics, Univ. of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews, Fife KY16 9 LZ, United Kingdom, tiago.marques@st-andrews.ac.uk), and Dave Moretti (NUWC, Newport, RI)

Dedicated passive acoustic density estimation studies with reliable survey design are still rare. We will present one such project: DECAF-TEA (Density Estimation for Cetaceans from Acoustic Fixed sensors in Testing and Evaluation Areas), which aims to estimate the density of beaked and fin whales on the SCORE US Navy range. We consider a mixed design that can deal with both species that produce acoustics signals with relatively short propagation distances (e.g., beaked whales) and those that produce signals with long propagation distances (e.g., fin whales), via distance sampling (DS) for the former and spatially explicit capture recapture (SECR) for the later. For beaked whales, we consider pairs of 3D bearing sensors to provide estimates of 3D localizations of detected whales, and from such data estimate detectability via distance sampling. At the same time, single units might provide a way to estimate density of fin whales via SECR. In either case, cue rates will be obtained from independent sources. In the context of DS we explore the impact of measurement error on the 3D bearing estimates and how this error propagates via errors in localization all the way through to potential bias in density estimation.

11:20

3aAB7. Song variability and change in the Australian pygmy blue whale song—Implications for passive acoustic abundance estimation. Capri D. Beck (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA 6102, Australia, capridawn.jolliffe@postgrad.curtin.edu.au), Alexander Gavrilov (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, WA, Australia), Robert McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA, Australia), Curt Jenner, and Micheline Jenner (Ctr. for Whale Res., Fremantle, WA, Australia)

Blue whales were heavily exploited throughout the early 20th century with many populations hunted to near extinction. Today, several known sub populations of blue whale exist, separated by geographic range and the acoustic signals they produce. The eastern Indian Ocean subpopulation of pygmy blue whales is easily identified by the production of a characteristic Australian song type. Long term collection of passive acoustic data from the Perth Canyon, Western Australia and Portland, Victoria, have revealed unprecedented levels of variability in the vocal behaviour of Australian pygmy blue whales. These levels of variability have significant implications for any attempts to assess the abundance of this population with acoustic techniques. Further, these findings may indicate higher levels of cognitive capability than previously expected in blue whales. The mechanisms behind variability in song production are unclear, though there is research to suggest that changes in vocal behaviour may be culturally driven or caused by changes in the ambient noise and environmental conditions. These findings highlight shortfalls in the current methods for measuring acoustic abundance, as well as indicate that higher than expected levels of variability may exist in the vocal behaviour of other pygmy blue whale subpopulations.

11:35

3aAB8. Learning deep features of spectrogram to estimate acoustic density of a fish species inhabitating *Posidonia oceanica* seagrasses. Lucas Lefèvre, Jean-Pierre Hermand, and Olivier Debeir (LISA - Environ. Hydro-Acoust. Lab, Université Libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels 1050, Belgium, lucas.lefevre@hotmail.com)

In field acoustic investigations on seagrass ecology, characteristic /kwa/ sound production of a fish species systematically present in meadows constituted by *Posidonia oceanica* species were recorded (Mediterranean, since 1995). Machine learning is investigated to automate estimation of acoustic density of the /kwa/ species. A multi-model deep neural network is implemented that directly inputs the spectrograms of signals recorded on hydrophones above and within the canopy. The network is a combination of 1D convolutional neural networks and a recurrent neural network with long short-term memory units and fully connected networks. Such architecture is able to process spectral information in the convolutional network and temporal information in the recurrent network, thereby making the best use of information in both time and frequency domains. In spite of high background noise level, the true-positive detection rate is shown to be excellent. With respect to human expert supervision, false-positive detections were, for the major part, faint sounds produced by distant fishes of the same species, as verified through careful re-listening. The apparent error rate is $8.4 \times 10-^{2}$ on a partially mislabeled dataset of three days (7-10 pm). The true error rate, after label correction, is estimated at $1\times 10-^{2}$.

Session 3aAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Arctic Acoustical Oceanography III

Peter F. Worcester, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225

Mohsen Badiey, Cochair
University of Delaware, University of Delaware, Newark, DE 19716

Hanne Sagen, Cochair
Polar Acoustic and Oceanography Group, Nansen Environmental and Remote Sensing Center, Thormøhlensgt 47, Bergen 5006, Norway

Contributed Papers

8:00

3aAO1. Measurements of underwater noise north of Spitsbergen using deep-sea recorders. Dag Tollefsen and Helge Buen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no)

This paper presents results from year-long passive acoustic recordings in the seasonally ice-covered ocean north of Spitsbergen (Svalbard archipelago) from July 2016 to June 2017. Two moorings were deployed by FFI, each with an AMAR Ultra Deep acoustic recorder (JASCO Applied Sciences) equipped with a single hydrophone. The moorings were deployed and retrieved during open-water conditions and remained during periods of partially to near fully ice-covered conditions. We present results from analysis of the acoustic data for ambient noise spectra and statistics, and discuss characteristics of the spectra in relation to environmental factors including ice cover, wind and ocean waves. Seasonal noise spectra are compared with historic measurements from the eastern Arctic. Components of the sound field including transients due to ice, marine mammals, and anthropogenic noise will also be discussed.

8:15

3aAO2. Short-term fluctuations in the ambient noise field at an Arctic chokepoint horizontal line array. Dugald Thomson, David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., Rm. 3635 - 1355 Oxford St., Halifax, NS B3H 4R2, Canada, dugald@dal.ca), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Garry J. Heard (Defence R&D Canada, Dartmouth, NS, Canada)

The Arctic is a region known for high variance in both ambient noise levels and local sound propagation conditions, and is currently experiencing an historic increase in shipping traffic, resulting in an acoustic environmental that is changing so rapidly that the sparse ambient noise data available from decades-old studies have been rendered obsolete. Renewed interest in Arctic acoustics have provided fresh noise data at northern sites, and in the summer of 2015 Defence R&D Canada collected a continuous two-week recording of the 48-hydrophone *Northern Watch* horizontal line array at Gascoyne Inlet. Over the ice-free study period, numerous ship passes, weather events, and anomalies were observed in the high-quality acoustic recordings. In this paper, spectral fluctuations, directionality, and correlation with ship tracks and weather observations in the noise data are analyzed to help improve the applicability of operational arctic noise models supporting sonar performance.

8:30

3aAO3. Modeling acoustic propagation in an ice-covered environment using a hybrid finite element-ray theory approach. Blake Simon and Marcia J. Isakson (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bsimon@arlut.utexas.edu)

Although models of acoustic propagation in an ice-covered environment have been calculated using only finite elements, ray theory models can cover longer distances and higher frequency ranges much more efficiently. In this study, a hybrid approach is taken by calculating the reflection coefficient of the ice using finite elements over a range of frequencies and angles. These values are then inserted into a ray-theory model. Using this model, a study of the sensitivity of common approximations of the ice cover on acoustic propagation is conducted. For example, the ice can be described simply as a pressure release surface or much more complexly as an elastic body with range and depth geo-acoustic property variations. The hybrid finite element-ray theory model will be verified numerically by comparing it to a full finite element propagation model for appropriate ranges and frequencies. [Work sponsored by ONR, Ocean Acoustics.]

8:45

3aAO4. Estimating channel capacity and sonar performance in the changing arctic. Paul C. Hines (Hines Ocean S&T Consulting, Dept. of Elec. and Comput. Eng., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Dainis Nams (GeoSpectrum Technologies Inc., Dartmouth, NS, Canada), Terry Deveau (JASCO Appl. Sci., Dartmouth, NS, Canada), Ron Kessel (Maritime Way Sci., Ottawa, ON, Canada), Jim Hamilton (JMH Consulting, Dartmouth, NS, Canada), Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada), and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Underwater acoustic processes including ambient noise, propagation, reverberation, and scattering have been studied for over half a century in the Canadian Arctic. Despite this, realistic predictions of communications and sonar performance have proved challenging due to the complex impact of ice cover on these processes, and the relative scarcity of sound speed profiles and information about the seabed. This challenge has been exacerbated in recent years because the rapid change in environmental conditions is reducing the relevance of many historical records. A key component of the present effort is to extract site specific model inputs from the environmental data contained in the literature, with an informed weighting toward more recent measurements. Site specific modelling was enhanced using inputs

from recent year-long ambient noise recordings. Model inputs were also influenced by specific source and receiver hardware being developed for use in future Arctic experiments. In this paper, low frequency sonar and communication performance estimates in the frequency band 20–250 Hz, based on site-specific environmental inputs and this low frequency hardware, are presented for several geographical regions of strategic relevance in the Canadian Arctic.

9:00

3aAO5. Low-frequency reverberation estimates using elastic parabolic equation solutions for ice-covered and ice-free Arctic environments. Scott D. Frank (Mathematics, Marist College, 3399 North Ave., Marist College Mathematics, Poughkeepsie, NY 12601, scott.frank@marist.edu) and Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A computational model of reverberation at low frequencies in an ice-covered environment is developed. The model is built on a full-field perturbation approach and includes elastic parabolic equation solutions for the acoustic field and its horizontal and vertical derivatives near water-ice and water-air interfaces. Our previous work demonstrated that average reverberation intensity is sensitive to both elasticity and thickness of the ice at mid-frequencies, where the ice layer thickness is on the order of or larger than both compressional and shear wavelengths. Here we address what happens to the reverberation when ice thickness approaches zero. Reverberation estimates for rough free surface and those from rough ice-water interface with increasing ice layer thickness are compared to determine at what ice thicknesses and acoustic frequencies the long-range reverberation distinguishes between the two cases. To isolate effects of ice thickness, we assume roughness is the same in both environments. Frequencies will be varied from very low, where the ice layer is practically transparent, to moderately low where the presence of the ice becomes noticeable in the reverberation. Numerical examples for reverberation in a typical Arctic environment with upward refracting sound-speed profile are presented and discussed. [Work supported by ONR.]

9:15

3aAO6. A parallelization of the wavenumber integration acoustic modelling package OASES. Gaute Hope (Polar Acoust. and Oceanogr., Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen 5006, Norway, gaute.hope@gmail.com) and Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

The wavenumber integration seismo-elastic model OASES can simulate the wave propagation in layered media, consisting of rough interfaces, elastic and porous layers. The wavenumber integration technique requires only the interesting frequencies, receiver depths, and offsets, to be calculated. As we seek to model pulse propagation at higher frequencies, denser sampling of frequencies must be applied. For complex media with many layers, and many vertical sections, the computation time quickly escalates. By exploiting that each frequency response can be calculated independently, a parallelization of the OASES package has been achieved and is presented here. This makes otherwise unfeasible or unpractical simulations computationally achievable. The parallel OASES package is applied to, and benchmarked on, several acoustic and seismic problems. The increased computation capacity is used to simulate and image the full wave field of several cases, reducing the computation time in one case from 1.5 years to 5 hours.

9:30

3aAO7. Inferring first-year sea ice thickness using broadband echosounders. Christopher Bassett (Resource Assessment and Conservation Eng., National Marine Fisheries Service, Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115, chris.bassett@noaa.gov), Andone C. Lavery (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jeremy P. Wilkinson (Br. Antarctic Survey, Cambridge, United Kingdom), Ted Maksym (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Zoe R. Courville (Cold Regions Res. and Eng. Lab., Hanover, NH)

Sea ice thickness is sparsely observed *in situ*. Under-ice acoustic techniques are a useful, but are generally used measure ice draft as a proxy for ice thickness. Using measurements of broadband acoustic backscattering (75–130 kHz) from laboratory-grown sea ice up to 80 cm thick, individual echoes from the water-ice and ice-air interfaces were isolated. Using the time delay between the echoes and a sound speed profile in the ice, total thicknesses were inferred and agree well with lengths of ice cores. This approach requires no ancillary data (i.e., water sound speed or pressure measurements). A temporal-domain model combining backscattering from both interfaces and from bubbles suspended within the brine channels also compares favorably with the measurements. This model is used to study the feasibility of applying the technique to first-year sea ice and to identify the bandwidths that best balance the constraints imposed by the scattering physics and practical considerations.

9:45

3aAO8. Measurements of sound speed in two types of ice with different microstructure. Anatoliy Ivakin, Wayne Kreider, and Bonnie Light (Appl. Phys. Laboratory, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

Measurements of sound speed were made on laboratory-grown ice samples serving as proxies for two types of sea ice with different microstructure. The first one, congelation ice, has relatively homogeneous structure typical for first-year sea ice. The second one, granular ice, with a more heterogeneous microstructure, was prepared as a proxy for retextured multi-year sea ice. Both samples had approximately equal bulk salinity (12 ppt) to isolate effects of different microstructure. Sound speed was estimated by measuring time of flight of 400 kHz pulses and was found in congelation ice to be about 10% higher than in granular ice. Error bounds were estimated to be within 1% based on same measurements made for plastic material with known sound speed. Therefore, the results confirm sensitivity of sound speed in ice to its internal microstructure and heterogeneity which are known to affect large-scale ice properties, such as strength, elasticity, permeability, air-sea exchange, habitability, and partitioning of shortwave solar radiation. This motivates future work that would include modeling and measurement of sound speed, attenuation and scattering in natural sea ice, with a goal to develop a remote acoustic sensing technique for sea-ice characterization and, in particular, discrimination between first-year and multi-year ice types.

10:00

3aAO9. Underwater bubble low frequency source for Arctic acoustic tomography, navigation, and communication. Andrey K. Morozov and Douglas C. Webb (Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

Arctic acoustic tomography requires a broadband source in the frequency range of 5–100 Hz. To meet this requirement, Teledyne Marine suggests applying a seismic source, recently developed for the Marine Vibrator Joint Industry Project. The source is considered to be less harmful for marine inhabitants and gives more precise and higher resolution imaging of the subsurface formations and structures. The coherent source is based on the application of an underwater, gas filled bubble resonator covered by an elastic membrane. The membrane supports high volume displacement. The sources are not sensitive to cavitation and to coupling effects. The fluid dynamics and acoustics of a spherical resonator are defined by the Rayleigh-Plesset equation. The buoyancy deforms the shape of a real bubble from spherical. The 3D simulation and experiments have shown that a cylindrical form is a practical engineering solution. It performances similar to a spherical bubble, keeps its shape and can be towed with a high speed. The Q-factor of a practical bubble resonator is ~10. Two systems are considered to cover a wide frequency band: a tunable resonator and a broadband multi-resonance system. The research proved that the bubble sound source is a practical solution for a frequency below 100 Hz.

Session 3aBAa

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

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 1K1, Canada

Chair's Introduction—8:15

Invited Papers

8:20

3aBAa1. Bone guided-wave ultrasonography: How far are we from clinical implementation? Lawrence H. Le, Tho N. Tran, Kim-Cuong T. Nguyen (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, 8440 - 112 St, Edmonton, AB T6G2B7, Canada, lawrence.le@ualberta.ca), and Mauricio D. Sacchi (Dept. of Phys., Univ. of AB, Edmonton, AB, Canada)

Bone guided-wave ultrasonography uses mechanical waves to study the long cortical bones. Long cortex has soft tissues above and marrow below acting like an ultrasound waveguide. The reverberations of the longitudinal and shear waves within the waveguide interfere constructively to generate energetic ultrasonic guided waves (UGW) travelling along the cortex. The UGW can be generated and recorded using an axial transmission technique with the transmitter and receiver deployed axially along the axis of the long bone on the skin's surface. The UGW thus acquired can be analyzed to provide information relevant to thickness and mechanical properties of the long bone. In this communication, we present an update of our research efforts on bone guided-wave ultrasonography including data acquisition, multichannel signal processing, bone modeling, and implementation of inversion algorithms to recover cortical thickness and mechanical parameters from UGW data. We also present some technical challenges, which will call for the joint effort of the bone ultrasound community to advance the science.

8:40

3aBAa2. High-resolution wavenumber estimation in ultrasonic guided waves using adaptive array signal processing for bone quality assessment. Shigeaki Okumura (Graduate School of Informatics, Kyoto Univ., Kyoto, Kyoto, Japan), Vu-Hieu Nguyen (Univ. of Paris-Est, Creteil, France), Hirofumi Taki (MaRI Co., Ltd., 22-44 Sakuragi-machi, Taihaku-ku, Sendai, Miyagi 982-0835, Japan, taki@marisleep.co.jp), Guillaume Haiat (CNRS, Creteil, France), Salah Naili (Univ. of Paris-Est, Créteil, France), and Toru Sato (Graduate School of Informatics, Kyoto Univ., Kyoto, Japan)

Quantitative ultrasound is a modality that is used to evaluate bone quality. It is considered that the analysis of ultrasound guided wave propagating along cortical bone may be useful for the assessment of cortical bone quality. Because the frequency-dependent wavenumbers reflect the elastic parameters of the medium, high-resolution estimation of the wavenumbers at each frequency is important. We report an adaptive array signal processing method with a technique to estimate the numbers of propagation modes at each frequency using information theoretic criteria and the diagonal loading technique. The proposed method estimates the optimal diagonal loading value required for guided wave estimation. We investigate the effectiveness of the proposed method via simple numerical simulations and experiments using a copper plate and a bone-mimicking plate, where the center frequency of the transmit wave was 1.0 MHz. An experimental study of 4 mm thick copper and bone-mimicking plates showed that the proposed method estimated the wavenumbers accurately with estimation errors of less than 4% and a calculation time of less than 0.5 s when using a laptop computer.

Contributed Papers

9:00

3aBAa3. Evaluation of axially transmitted shear waves in cortical bone: Experimental and simulation study. Leslie V. Bustamante Diaz, Masaya Saeki, and Mami Matsukawa (Dept. of Elec. and Electronics Eng., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, cyjb3302@mail4.doshisha.ac.jp)

Quantitative ultrasound has become a useful technique for bone assessment. Axial transmission technique has demonstrated various advantages over the through-transmission technique. Longitudinal wave and Lamb modes measurements in bone have been extensively reported while shear waves have been neglected. Shear wave velocities are lower than longitudinal wave velocities and can be useful for bone torsional strength characterization. In this study, plate samples were fabricated from bovine cortical bone. An ultrasonic pulse was transmitted along the long axis of the sample and shear waves were experimentally detected in the MHz range. Using time-of-flight technique, velocities were determined considering three different points of the received wave as a function of the ultrasound incident angle, transmitter-receiver distance and bone plate thickness. The observed shear wave velocities varied in a range (1630 to 1860 m/s) depending on the incident angle and the particular wave path, due to the anisotropy and heterogeneity of the bone structure. A 2D simulation using FDTD method was developed to understand the wave propagation phenomenon. Experimental and simulation results showed good agreement respect to the shear wave velocity estimated in a through-transmission measurement (1720 m/s), specially in angles close to the critical angle (60°).

9:15

3aBAa4. Model-based inversion of ultrasonic guided waves for cortical bone properties. Tho N. Tran (Dept. of Radiology and Diagnostic Imaging, Univ. of Alberta, Res. Transition Facility, Edmonton, AB T6G 2V2, Canada, tho.tran@ualberta.ca), Mauricio D. Sacchi (Dept. of Phys., Univ. of AB, Edmonton, AB, Canada), Dean Ta (Dept. of Electron. Eng., Fudan Univ., Shanghai, China), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multi Echelle UMR 8208 CNRS, Université Paris-Est, Creteil, France), and Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, Edmonton, AB, Canada)

Axial transmission ultrasonography, which uses a set of transmitting and receiving probes placed on the same waveguide's surface, shows the potential clinical application for cortical bone quality assessment. In this work, a model-based parameter sweep inversion approach has been developed to estimate the thickness and elastic velocities of the cortex from the dispersive axially-transmitted snapshots. The inversion algorithm is formulated in the frequency-phase velocity (*f-c*) domain. To solve the inverse problem, i.e., to extract bone properties from ultrasound data, a forward modeling has been developed to simulate the *f-c* dispersion curves given a bone model. A semi-analytical finite element (SAFE) method is used to compute the dispersion curves for a complex structure of a cortical bone plate coupled with overlying soft tissues. A parameter sweep is used to seek within a range of values an optimized solution with the least misfit. The proposed method optimizes the mismatch between the measured and theoretically calculated dispersion curves with a least-square constraint. Numerical and *in-vivo* experimental data examples are presented to illustrate the technique's performance.

9:30

3aBAa5. Inverse characterization of radius bone using low-frequency ultrasonic guided waves. Daniel Pereira (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, Université de Montreal, Montreal, QC, Canada), and Pierre Belanger (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

The success of the cortical bone characterization using the axial transmission is highly dependent on the inversion model used to match experimental data with the true bone properties. Simplified models such as plate or cylinder are typically used in the literature. In our previous work, a more elaborate model based on a bone-like geometry built using semi-analytical finite element (SAFE) method was introduced. The model was successfully implemented in an inverse characterization routine using laboratory-controlled measurements on bone phantoms. Thus, the aim of this work is to apply the proposed inverse scheme on the characterization of *ex-vivo* radius samples. In order to do so, five radiuses were taken from donors aged between 53 and 88 years old and tested using typical axial transmission configuration. Ultrasonic guided wave modes were excited at low-frequency (100-400 kHz) using a piezoelectric transducer and measured using a laser interferometer at the middle of the diaphysis. The measured velocities were systematically compared to velocities obtained with the SAFE model in order to predict the cortical bone properties. For each sample, four parameters were estimated: (1) Young's modulus; (2) density; (3) thickness; and (4) outer diameter. The results showed a notable correlation of the thickness and outer diameter with respect to the μCT images of the samples, while a less significant correlation was observed for the Young's modulus and density with respect to the gray level of the μCT images.

Invited Papers

9:45

3aBAa6. Effect of scattering and porosity on the apparent absorption in porous biphasic structures—Application to cortical bone. Yasamin Karbalaeisadegh, Omid Yousefian, and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

In this in-silico study, the attenuation coefficient in porous structures mimicking simplified cortical bone is calculated both in presence and absence of absorption, to isolate the effect of scattering on attenuation. Pore distributions with various pore concentrations (5–25 pore/mm^2) and diameters (40–120 μm) are generated. The propagation of 8 MHz plane waves is simulated using a 2D FDTD package. The attenuation coefficient is measured by fitting the exponential decay of the signal amplitude in the frequency domain during propagation in the absence or presence of absorption (coefficient of 10 dB/cm/MHz to mimic bone tissue), for different pore concentrations and sizes. The difference between total attenuation and scattering attenuation coefficients (obtained with no absorption) is defined as the apparent absorption coefficient. For ka<1 (k: wave number; a: scatterer diameter) the effect of absorption does not change with increasing porosity. However, for ka>1, increasing porosity leads to a higher apparent absorption, indicating a higher impact of absorption, even though the absorption coefficient and the overall volume fraction of absorbing material (solid phase) were kept constant. This suggests complex interactions between the wave and the structure, and could be attributed to longer propagation paths in the solid phase due to multiple scattering by the pores.

10:05–10:20 Break

Contributed Paper

10:20

3aBAa7. Characterizing porosity in bone-like porous media from frequency dependent attenuation. Rebekah White (Ctr. for Res. in Sci. Computation, North Carolina State Univ., Chapel Hill, NC), Omid Yousefian (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), H. T. Banks (Ctr. for Res. in Sci. Computation, North Carolina State Univ., Raleigh, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Osteoporosis affects both pore size and density in cortical bone. Quantifying levels of osteoporosis by inferring these micro-architectural properties from ultrasonic wave attenuation in cortical bone has yet to be done. Here, we propose a phenomenological power law model that captures the dynamics of frequency dependent attenuation in non-absorbing porous media mimicking a simplified cortical bone structure. We first demonstrate that, although it is frequency dependent, apparent absorption does not depend upon pore density and pore concentration, justifying the use of non-absorbing media for the simulations. We generate scattering attenuation data for various combinations of pore diameters (ranging from 20 to 100 μm) and pore densities (ranging from 3 to 10 pore/[mm]^2) using a 2D FDTD package (Simsonic), which simulates the propagation of elastic waves at frequencies of 1–8 MHz. The model is then optimized to fit these datasets by solving an inverse problem under an ordinary least squares (OLS) framework. With this we establish linear, functional relationships between the optimized power law model parameters and the micro-architectural parameters. These relationships showed that ranges of porosity could be inferred from attenuation data. Applying these techniques to attenuation data from cortical bone samples could allow one to characterize the micro-architectural properties of bone porosity.

Invited Papers

10:35

3aBAa8. Numerical simulation and machine learning based analysis of the ultrasonic waveform propagating 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), Shigeaki Okumura, and Shuqiong Wu (Graduate School of Informatics, Kyoto Univ., Kyoto, Kyoto, Japan)

Since bone has a complex structure, it is difficult to analytically understand the behavior of the ultrasound although it is useful for bone quality diagnosis. Our group, therefore, had been working on simulating ultrasound propagation inside bone models. In this paper, the results of the neural network based bone analysis using waveforms derived by the 3-D elastic FDTD (finite-difference time domain) simulation will be presented as well as its basis and some representative results of the FDTD method applied for the bone assessment. Since the FDTD simulation only requires the 3-D geometry of the model and the acoustic parameters (density, speed of longitudinal wave and shear wave) of the media, it has been very useful for evaluating each effect of a certain acoustic parameter or the bone geometry such as bone density, respectively, in addition to the purpose of visually understanding the wave behavior inside the model including the propagation path. Moreover, thanks to the recent powerful computational resource, it is now realized to prepare a huge number of waveforms for machine learning by using FDTD method. As a result, it was shown that the neural network can estimate the bone density better than the traditional method. (Grant: JSPS KAKENHI 16K01431.)

10:55

3aBAa9. Detection of fast and slow waves propagating through porous media. 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)

Porous media can support two longitudinal waves, which often overlap in time and frequency domains. Each wave has its own attenuation coefficient and phase velocity. These properties are related to volume fraction of solid phase, tortuosity, viscous characteristic length, and elasticity. Therefore, knowledge of individual wave properties is useful for characterizing porous media. Accordingly, methods for recovering separate reconstructions of fast and slow waves are of interest. The transfer function of a porous sample may be expressed as a weighted sum of two complex exponentials. The Modified Least Squares Prony's (MLSP) method may be used to recover the two individual components for non-dispersive media. Porous media are dispersive, however. The MLSP method may be augmented with curve-fitting (MSLP + CF) to account for dispersion. An alternative approach, based on Bayesian probability theory, is also powerful for recovering fast and slow waves. These approaches have been tested in through-transmission experiments on cancellous bone samples. Bayesian and MLSP + CF approaches successfully separated fast and slow waves and provided good estimates of the fast and slow wave properties even when the two wave modes overlapped in both time and frequency domains.

11:15

3aBAa10. Cancellous bone characterization using ultrasonic backscatter parametric imaging. Dean Ta, Ying Li (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn), Boyi Li (Dept. of Electron. Eng., Fudan Univ., Shanghai, Shanghai, China), Rui Zheng (School of Information Sci. and Technol., ShanghaiTech Univ., Shanghai, China), and Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of Alberta, Edmonton, AB, Canada)

The ultrasonic imaging of the highly attenuated porous media (like spongy bone) is of great interest to the researchers. In this study, we used a 7.5 MHz focal transducer (V321, Panametrics, USA) to scan a total of 35 bovine cancellous bone samples. The −-6dB lateral resolution of the transducer was 0.157 mm which is comparable to the trabecular bone thickness (0.1–0.2mm). Apparent integrated backscatter (AIB) was calculated using different signal of interest (SOI) to reconstruct the ultrasonic backscatter parametric images. The trabecular bone structure can be clearly seen from the images. AIB was positively correlated with bone mineral density (BMD) ($R = 0.61$, $p<0.05$), bone volume fraction (BV/TV) ($R = 0.53$, $p<0.05$), and apparent bone density (ABD) ($R = 0.58$, $p<0.05$) when SOI was chosen around the surface of the bone sample, but negatively correlated with BMD ($R=-0.77$, $p<0.05$), BV/TV ($R=-0.81$, $p<0.05$), and ABD ($R=-0.82$, $p<0.05$) when SOI was chosen below the surface of the bone sample. We suppose that single scattering (SS) and multiple scattering (MS) may lead to this opposite correlation phenomena. When wavelength is comparable to the scatterer diameter, diffractive scattering is complex as predicted by Faran cylinder model. We suggest that MS may play an important role in cancellous bone characterization.

Contributed Paper

11:35

3aBAa11. Ultrasonic backscatter evaluates the variability of bone in pregnant women. Boyi Li, Dan Li, and Dean Ta (Dept. of Electron. Eng., Fudan Univ., Fudan University, Handan Rd. 220, Shanghai, 200433, China, Shanghai, Shanghai 200433, China, 16110720100@fudan.edu.cn)

Pregnancy and lactation-associated osteoporosis (PLO) is characterized by the deterioration of bone tissue and the fractures during pregnancy without any underlying disorders. This study attempted to investigate the variability of bone in pregnant women. The ultrasonic backscatter experiments were performed in Shanghai Changning Maternity & Infant Health Hospital. A total of 896 subjects were divided into four groups gestational age from 11^{th} to 39^{th} week, calculated apparent integrated backscatter coefficients (AIB) and spectral centroid shift (SCS). The estimated bone mineral density (Est. BMD) was measured using a medical ultrasonic transmission instrument. The body mass index (BMI) was calculated basing on height and weight. The results showed that AIB preformed a decreasing trend with the increasing of gestational age, the mean value in 37-39^{th}-week group decreased 2.07 dB comparing with the 11-13^{th}-week group, $p<0.001$. The SCS mean value in 37–39^{th}-week group shifted 21.66 kHz than the 11–13^{th}-week group, $p<0.001$. The Est. BMD mean value in 37–39^{th}-week group decreased 0.05 g/cm^2 than the 11–13^{th}-week group, $p<0.001$. The BMI increased with gestation age, the mean value in the 37–39^{th}-week group was higher than the 11–13^{th}-week about 4.19 kg/m^2, $p<0.001$. One of the reasons for these results was that pregnant women provide nutrition for the fetus, resulting in decreased bone tissue along with gestational age increasing. These results suggest that ultrasonic backscatter method might have a potential to diagnosis PLO.

WEDNESDAY MORNING, 7 NOVEMBER 2018 SIDNEY (VCC), 9:00 A.M. TO 11:50 A.M.

Session 3aBAb

Biomedical Acoustics and Physical Acoustics: Bubble Trouble in Therapeutic Ultrasound I

Klazina Kooiman, Cochair
Thoraxcenter, Dept. of Biomedical Engineering, Erasmus MC, P.O. Box 2040, Room Ee2302, Rotterdam 3000 CA, Netherlands

Christy K. Holland, Cochair
Internal Medicine, Division of Cardiovascular Diseases and Biomedical Engineering Program, University of Cincinnati, Cardiovascular Center Room 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Invited Paper

9:00

3aBAb1. Microbubble gas volume: A unifying dose parameter in blood-brain barrier disruption by focused ultrasound. Kang-Ho Song (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., campus box 427, Boulder, CO 80309), Brandon Harvey (Neurosci., NIH Bayview Res. Ctr., Baltimore, MD), and Mark Borden (Mech. Eng., Univ. of Colorado, Boulder, CO, mark.borden@colorado.edu)

Focused ultrasound with microbubbles is being developed to transiently, locally and noninvasively disrupt the blood-brain barrier (BBB) for improved pharmaceutical delivery. Prior work has demonstrated that, for a given concentration dose, microbubble size affects both the intravascular circulation persistence and extent of BBB opening. In this study, we independently measured the effects of microbubble size (2 vs. 6 μm diameter) and concentration, covering a range of overlapping gas volume doses (1–40 μL/kg). We first demonstrated precise targeting and a linear dose-response of Evans Blue dye extravasation to the rat striatum for a set of constant microbubble and ultrasound parameters. We found that dye extravasation increased linearly with gas volume dose, with data points from both microbubble sizes collapsing to a single line. A linear trend was observed for both the initial sonication and a second sonication on the contralateral side. Based on these results, we conclude that microbubble gas volume dose, not size, determines the extent of BBB opening by focused ultrasound (1 MHz, ~0.5 MPa at the focus). Finally, using optimal parameters determined for Evan blue, we demonstrated gene delivery and expression using a viral vector one week after BBB disruption.

Contributed Papers

9:20

3aBAb2. Persistent Pests: The role of gas diffusion in histotripsy bubble longevity. Kenneth B. Bader and Viktor Bollen (Dept. of Radiology, Univ. of Chicago, 5835 S. Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

Histotripsy is a focused ultrasound therapy under development for tissue ablation via the mechanical action of bubble clouds. While strong bubble activity is a hallmark of histotripsy, persistent bubble clouds shield the focal zone from subsequent pulses and prevent complete liquefaction of the target tissues. The diffusion of gas into the bubble during histotripsy exposure may be a contributing factor to the longevity of histotripsy-induced cavitation. To investigate the role of gas diffusion in bubble persistence, the oscillations of a single nanoscale nucleus exposed to a histotripsy pulse were computed in parallel with a first-order diffusion equation. The bubble gas content increased more than five orders of magnitude during the expansion phase, inflating the equilibrium bubble diameter by more than a factor of 50. The inertial collapse of the gas-filled bubble was arrested significantly in comparison to computations neglecting diffusion, as indicated by the minimum bubble size and maximum bubble wall speed during the collapse. The residual gas bubble required more than 15 ms for passive dissolution post excitation, consistent with experimental observation. These results indicate gas diffusion does not influence the bubble dynamics during the excitation, but is a contributing factor to bubble persistence between histotripsy pulses.

9:35

3aBAb3. Effect of high intensity focused ultrasound transducer F-number and waveform non-linearity on inertial cavitation activity. Christopher Bawiec (School of Medicine, Div. of Gastroenterology, Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98103, bawiec@uw.edu), Christopher Hunter, Wayne Kreider (CIMU, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Vera A. Khokhlova, Oleg A. Sapozhnikov (CIMU, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Tatiana D. Khokhlova (School of Medicine, Div. of Gastroenterology, Univ. of Washington, Seattle, WA)

Enhanced chemotherapeutic drug delivery has been shown using pulsed high intensity focused ultrasound (pHIFU) without contrast agents. The threshold of these inertial cavitation effects has recently been correlated to the formation of shocks at the focus. The shock amplitude and corresponding peak negative pressure (p-) are primarily determined by the transducers F-number with less focused (higher F-number) transducers producing shocks at lower p-. Here, the dependence of inertial cavitation activity on F-number was investigated in gel phantoms using passive cavitation detection (PCD) and high-speed photography. HIFU transducers with the same aperture but different F-numbers (0.77, 1.02, and 1.52), operable at three driving frequencies (1 MHz, 1.5 MHz, and 1.9 MHz), were utilized with driving conditions consisting of 1 ms pulses delivered every second and p- from 1 to 15 MPa. Broadband noise emissions recorded by PCD were batch-processed to extract cavitation probability and persistence while concurrent imaging was performed in the focal pHIFU region. At the same p-, both PCD metrics and the imaging revealed enhanced cavitation activity at higher F-numbers. These results support the use of less focused, smaller-footprint transducers for achieving desired cavitation-aided drug delivery. [Work supported by NIH R01EB023910, K01DK104854, R01EB007643, and RFBR 17-54-33034.]

9:50

3aBAb4. Targeted microbubble opening of cell-cell junctions for vascular drug delivery elucidated with combined confocal microscopy and Brandaris 128 imaging. Ines Beekers, Merel Vegter, Kirby R. Lattwein, Frits Mastik, Robert Beurskens, Antonius F. W. van der Steen (Biomedical Eng., Erasmus MC, Office Ee2302, P.O. Box 2040, Rotterdam, Zuid Holland 3000 CA, Netherlands, d.beekers@erasmusmc.nl), Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands), Nico de Jong (Acoust. Wavefield Imaging, Delft Univ. of Technol., Rotterdam, Netherlands), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

Ultrasound insonification of microbubbles enhances vascular drug delivery pathways, such as opening of cell-cell junctions and pore formation (sonoporation). However, the underlying mechanism remains unknown. The aim of our study was to elucidate the microbubble-cell interaction using the Brandaris 128 ultra-high speed camera (~17 Mfps), to visualize microbubble oscillation, coupled to a custom-build Nikon confocal microscope, to visualize cellular response. Confluent endothelial cells were evaluated for opening of cell-cell junctions with Cell Mask and for sonoporation with Propidium Iodide (PI). The cellular response of single targeted microbubbles ($n = 168$) was monitored up to 4 min after ultrasound insonification (2 MHz, 100–400kPa, 10-cycles). Cell-cell junctions opening occurred more often when cells were only partially attached to their neighbors (45%) than when fully attached (15%). Almost all fully attached cells showing cell-cell opening also showed PI uptake (92%). The mean microbubble excursion was larger when a cell was sonoporated (1.0μm) versus non-sonoporated (0.47μm). Additionally, larger microbubble excursion amplitudes correlated with larger pore size coefficients, obtained by fitting the PI uptake profiles to the Fan model [Fan, *et al.*, PNAS, 2012]. In conclusion, using the state-of-the-art imaging system we can now elucidate the relationship between microbubble oscillation behavior and the drug delivery pathways.

10:05–10:20 Break

10:20

3aBAb5. Prototypical madness: Design and demonstration of a magnetic-acoustic targeted drug delivery system. Michael Gray, Estelle Beguin, Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, IOxford OX37DQ, United Kingdom, michael.gray@eng.ox.ac.uk), Heather Nesbitt, Keiran Logan, Sukanta Kamila, Anthony McHale, John Callan (School of Pharmacy and Pharmaceutical Sci., Ulster Univ., Coleraine, United Kingdom), and Lester Barnsley (Heinz Maier-Leibnitz Zentrum, Jülich Ctr. for Neutron Sci., Julich, Germany)

Ultrasound, microbubbles, and magnetic nanoparticles have been used both separately and in varying combinations for targeted drug delivery. Recent studies have demonstrated the therapeutic benefit of magnetic microbubble (MMB) retention and acoustic targeting using separate devices. As a developmental step towards clinical implementation, a magnetic-acoustic device (MAD) was designed for the purpose of generating co-aligned magnetic and acoustic fields with a single hand-held enclosure. This paper presents *in vitro* characterization and *in vivo* demonstration of a targeted therapeutic system wherein the MAD non-invasively retains and activates drug-loaded MMBs. Free field experiments were conducted in order to characterize the magnetic field and its gradient for MMB capture, and to quantify acoustic field strength and directivity. Flow phantom experiments were used to quantify MMB retention and illustrate the resulting enhancement of cavitation activity. Murine experiments then demonstrated therapeutic efficacy in a pancreatic cancer model, showing a significant benefit in comparison to the use of separate magnetic and ultrasonic devices.

10:35

3aBAb6. Feeling gassy? Modifying the oxygen partial pressure of a fluid using acoustic droplet vaporization and different droplet concentrations. Haili Su, Karla P. Mercado-Shekhar, Rachel P. Benton (Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209), Bin Zhang (Div. of Biostatistics and Epidemiology, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Sneha Sharma, and Kevin J. Haworth (Univ. of Cincinnati, Cincinnati, OH, kevin.haworth@uc.edu)

Nucleating acoustic droplet vaporization reduces the dissolved gas content in a fluid. The objective of this study was to determine if the change in the partial pressure of oxygen (Po_2) could be modulated by adjusting the concentration of micron-sized perfluoropentane droplets. The droplets were manufactured using high-speed shaking and size-isolated using differential centrifugation (1 to 6 μm in diameter). Droplets were diluted in saline with a Po_2 of 154 mmHg and pumped through a 37°C flow phantom at 40 mL/min. The concentration ranged from 3.5×10^6 to 3.5×10^8 droplets/mL. A 5 MHz focused transducer insonified droplets at peak negative pressures of 4.05 MPa with a 500-Hz pulse repetition frequency and 5-cycle pulse duration. The Po_2 was measured downstream of the insonation region. At the lowest droplet concentration, the Po_2 was reduced to 129 mmHg. As the droplet concentration was increased, the Po_2 was reduced further. The reduction was in agreement (intra-class correlation and Pearson correlation coefficients greater than 0.9) with the model reported by Radhakrishnan *et al.* (2016, Ultrason. Sonochem.). At the highest droplet concentration, the Po_2 was reduced to 31 mmHg. These results demonstrate that ADV with varying droplet concentration modulates the oxygen partial pressure in a fluid.

10:50

3aBAb7. Phospholipid-coated microbubbles: Controlling response to ultrasound. Simone A. Langeveld (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rotterdam 3000 CA, Netherlands, s.a.g.langeveld@erasmusmc.nl), Ines Beekers, Antonius F. W. van der Steen, Nico de Jong, and Klazina Kooiman (Biomedical Eng., Erasmus MC, Rotterdam, Zuid Holland, Netherlands)

Variability in response to ultrasound is an issue when microbubbles are used for drug delivery. This may be caused by immiscible phospholipid components and lipid phase separation in microbubble coatings. Since cholesterol influences phase distribution in lipid monolayers, we studied its effect (10 mol. %) on DSPC-based microbubbles. Lipid phase and ligand distribution of microbubble coatings were studied using super-resolution microscopy. Microbubble behavior upon ultrasound insonification (20 to 50 kPa, 2 MHz, single 10-cycle burst) was studied using the Brandaris 128 ultra-high-speed camera. As expected, lipid phase separation was observed in microbubble coatings without cholesterol. However, in microbubbles with cholesterol no lipid phase separation was observed. Without cholesterol, the ligand distribution was heterogeneous for only 3.9% of the microbubble surface (n = 42), while this was significantly more (11.6%) for microbubbles with cholesterol (n = 25), likely due to buckles. Moreover, cholesterol-containing microbubbles retained characteristic resonance behavior in response to ultrasound. These results demonstrate that although cholesterol in the microbubble coating affects the miscibility and ligand distribution of phospholipid components, a functional response to ultrasound is preserved. The phospholipid-cholesterol-coated microbubbles we developed may therefore have potential as theranostic agents. [Funding by the Phospholipid Research Center in Heidelberg, Germany, is gratefully acknowledged.]

11:05

3aBAb8. Multi-source passive acoustic source localisation. Catherine Paverd, Erasmia Lyka, and Constantin Coussios (Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, catherine.paverd@eng.ox.ac.uk)

Accurate acoustic source localisation (ASL) has significant potential to improve diagnostic ultrasound imaging of small vessel structures. Recently, super-resolution imaging techniques for both active and passive source mapping have been developed; however, these approaches assume the presence of a single source within the point spread function (PSF) of the system. In reality, multiple sources may be present, for example, when high concentrations of contrast agent are used. We use a two-step approach to localise multiple sources within the PSF of a clinically relevant passive cavitation detection system. First, we apply a Blind Source Separation (BSS) technique known as Independent Component Analysis, which relies on higher-order statistics to separate and reconstruct signals originating from independent sources. Second, we determine the time-difference-of-arrival of the separated signals on each receiver, and perform ASL by fitting a polynomial using least squares regression. Simulation and experimental results demonstrate that with the BSS-ASL combination, multiple sources aligned axially within the PSF of a single array located 70 mm from the focus can be localised with sub-millimetre resolution. We verify sources as distinct using Passive Acoustic Maps generated from a perpendicular linear array. Further work is necessary to ascertain performance limits and to validate the technique *in vivo*.

11:20

3aBAb9. Non-linear acoustic emissions from therapeutically driven microbubbles. Paul Prentice and Jae Hee Song (CavLab, Medical and Industrial Ultrason., Univ. of Glasgow, Ninewells Hospital, Dundee DD1 9SY, United Kingdom, paul.prentice@glasgow.ac.uk)

Acoustic detection of contrast agent microbubbles, infused to the vasculature for exposure to focused ultrasound, is now routinely undertaken to evaluate therapy and avoid irreversible tissue damage. Harmonic, subharmonic (to the frequency of the focused ultrasound, f_0) and broadband emissions are often used to distinguish between stable and inertial cavitation activity, and associated bioeffects. The driven microbubble dynamic responsible for the generation of non-linear emissions, however, may not be well understood. Results from an investigation of single SonoVue microbubbles flowing through a capillary, for exposure to focused ultrasound at f_0 = 692 kHz, will be presented. Dual high-speed imaging from orthogonal perspectives at circa 2×10^5, and shadowgraphically at 10 million frames per second, capture microbubble activity and shock wave generation. Acoustic emissions are simultaneously collected with a calibrated broadband needle hydrophone, and high frequency imaging array on passive receive, for cavitation mapping. The results indicate that non-linear emissions are mediated by periodic bubble collapse shock waves, with subharmonic emission occurring above a threshold driving pressure amplitude. Implications for detection and quantification of driven microbubble cavitation activity, as well as conventional classification as stable or inertial, will be discussed. [This work was supported by ERC Starting Grant TheraCav, 336189.]

11:35

3aBAb10. Characterization of lipid-encapsulated microbubbles for delivery of nitric oxide. Himanshu Shekhar, Arunkumar Palaniappan (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Cameron McDaniel, Daniel J. Hassett (Dept. of Molecular Genetics, Biochemistry & Microbiology, Univ. of Cincinnati, Cincinnati, OH), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Nitric oxide (NO) is a potent bioactive gas capable of inducing vasodilatory, anti-inflammatory, and bactericidal effects. The short half-life, high reactivity, and rapid diffusivity of NO make therapeutic delivery challenging. The goal of this work was to develop a technique to sequester NO within lipid-shelled microbubbles. Microbubbles loaded with either NO alone (NO-MB) or with NO and octafluoropropane (NO-OFP-MB) were synthesized by high-shear mixing of 1 mL lipids with either 1 mL of NO, or a mixture of 0.9 mL NO and 0.1 mL OFP. The size distribution and attenuation coefficient of NO-MB and NO-OFP-MB were measured using a Coulter counter and a broadband acoustic attenuation spectroscopy system, respectively. The payload of NO in the microbubbles was assessed using an amperometric micro-electrode sensor. Co-encapsulation of NO with OFP increased the number density, attenuation coefficient, and temporal stability of lipid-shelled microbubbles. However, the amount of NO loaded in NO-MB and NO-OFP-MB was similar (0.91 $\pm$ 0.03 μM and 0.93 $\pm$ 0.1 μM, respectively). These results suggest that NO can be encapsulated within lipid-shelled microbubbles. However, co-encapsulation of OFP does not enhance the NO payload or the temporal stability, despite an increase in attenuation and number density of lipid-shelled microbubbles.

Session 3aED

Education in Acoustics: Hands-On Demonstrations

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

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 Victoria. 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 effor 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, 7 NOVEMBER 2018 CRYSTAL BALLROOM (FE), 9:45 A.M. TO 12:00 NOON

Session 3aMU

Musical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Percussion Instruments

Uwe J. Hansen, Cochair
Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Andrew C. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Invited Papers

9:45

3aMU1. The state of percussion research in musical acoustics: How we got here and where it is going. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

Drums are among the most ancient of all musical instruments and have been found in nearly all cultures across the world. This work covers a selection of major investigations of the acoustics of percussion instruments by a variety of scientists throughout history. Ernst Chladni's efforts to characterize the vibration of plates are well known in acoustics and often replicated by students in the laboratory. Other scientists who made early contributions to the understanding of percussion instruments include Lord Rayleigh and C. V. Raman. Rayleigh's observations of the acoustics of the kettledrum and bells represented only a fraction of his contributions to acoustics. Raman, who is remembered more for his work in spectroscopy, made detailed studies of the traditional Indian drum, the tabla. Modal analysis techniques used to investigate the acoustics of percussion instrument have evolved significantly in recent years. A brief review of current modal analysis techniques for studying percussion instruments is presented in this work. In addition, suggestions for future directions of percussion research is described here.

10:05

3aMU2. Musical drumhead damping using externally applied commercial products. Randy Worland and William Miyahira (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Snare drums and tom toms used in drum sets often produce an undesirable high frequency ringing sound when struck. Drummers have historically applied DIY damping solutions to address this ringing, placing objects such as wallets and tape on the heads. A large variety of inexpensive commercial products is now available to dampen drums in a more controlled manner. The most commonly used products consist of small adhesive pads that can be placed directly on the drumhead at desired locations, either singly or in combinations. An experimental investigation of the ringing drumhead modes and the effects of the damping pads is reported. Decay rate measurements indicate that the (3,1), (4,1), and (5,1) modes often contribute the most to the ringing sound, and that these modes are strongly affected by the damping pads. Another class of commercial dampener consists of thin annular rings of Mylar that conform to the perimeter of the drum head. These free-floating rings produce a similar damping effect, but dissipate energy through a different physical mechanism. Experimental results for both types of dampeners are presented and discussed in terms of viscoelastic and air-layer damping mechanisms, respectively.

10:25

3aMU3. Steel pan developments. Uwe J. Hansen (Phys., Utah Valley Univ., 6104 N Lake Mt Rd., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu) and Thomas Rossing (CCRMA, Stanford Univ., Los Altos Hills, CA)

The Caribbean steel pan is likely the single most significant new acoustic musical instrument of the 20th Century. Some major developments incorporated by Felix Rohner of Panart have led to important instrument modifications. Among the differences to be discussed are as follows: using specified steel alloys in the sheet metal to replace commercial 55 Gal drums, sinking the pan in a press rather than by hand, surface hardening the playing surface in a nitride bath, dispensing with chiseled note section boundaries, adding a central dome to each note section, replacing the pan by a Hang (a lap-held instrument played by hand), and finally an additional air volume enclosed, to enhance low frequency resonances (the Gubal).

10:45–11:05 Break

11:05

3aMU4. Take me out to the ballgame: The acoustics of the softball bat piano. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

As a multimedia supplement for a recent article "Acoustics and Vibration of Baseball and Softball Bats" [*Acoustics Today*, **13**(4), 35–42 (2017)] the author created a "softball bat piano" and played the tune "Take Me Out to the Ballgame." The bat piano is a collection of softball bats, clamped at the handles, selected for their barrel frequencies. It is played by striking the bat barrels with a softball. The structural modes of the hollow cylindrical barrel of an aluminum or composite bat are responsible for the trampoline effect which affects the performance of the bat as it is used for the game of softball. However, the "hoop mode" which gives rise to the trampoline effect is also the source of the bat piano's audible musical sound. The nature of these cylindrical shell vibrations will be discussed, highlighting similarities to the musical properties of bells and chimes. The influence of bat construction and materials on the "hoop frequency" will be explained, along with the process of selecting bats with a range of frequencies wide enough to form a musical scale. If time permits, video of additional musical selections will be shown.

11:25

3aMU5. How beatboxers produce percussion sounds: A real-time magnetic resonance imaging investigation. Timothy Greer (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA 90007, timothdg@usc.edu), Reed Blaylock (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA), Nimisha Patil, and Shrikanth S. Narayanan (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA)

Beatboxing is a musical artform in which performers use their vocal tract to create percussion sounds. Sometimes beatboxers perform as a part of an ensemble, using their vocal tract to provide a beat for other musicians; other times, beatboxers perform alone, where they might sing and produce percussion sounds simultaneously. We present methods in real-time magnetic resonance imaging (rtMRI) that offer new ways to study the production of beatboxing sounds. Using these tools, we show how beatboxers can concatenate intricate articulations to create music that mimics the sound of percussion instruments and other sound effects. The rtMRI methodology reveals how different beatboxers play their vocal folds to perform characteristic "clean" or breathy styles. By using rtMRI to characterize different beatboxing styles, we show how video signal processing can demystify the mechanics of artistic style.

Contributed Paper

11:45

3aMU6. Preliminary modal analysis of the Alo Gong. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com), Tracianne B. Neilsen (Brigham Young Univ., Provo, UT), and Maria T. Keke (Music, Univ. of Nigeria, Nsukka, Nigeria)

The Alo is a percussion instrument found predominantly in the eastern parts of Nigeria with the Igbo's and plays an important role in the Igbo musical culture and settings. Being a bass gong with tones of three pitches, the Alo produces a very distinct sound and a sustained reverberation rendering a background beat to music of an African instrument ensemble or a mixed ensemble. This paper discusses the acoustical output characteristics of this instrument. The spectral characteristics of the sustained reverberation is presented, along with the dynamics responses of the structure to an excitation. These analyses not only provide an understanding of the distinct sound of the Alo instrument but may also be used by African instrument acoustician to re-model, or create more instruments with resonance frequencies.

Session 3aNS

Noise, ASA Committee on Standards, and Signal Processing in Acoustics: Technological Challenges in Noise Monitoring

Matthew G. Blevins, Cochair
U.S. Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822

Anton Netchaev, Cochair
U.S. Army Engineer Research and Development Center, Vicksburg, MS

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. Benefits and challenges of using consumer audio equipment for unattended acoustical monitoring. Daniel J. Mennitt (Elec. and Comput. Eng., Colorado State Univ., 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), Damon Joyce, and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

The quality of an acoustical measurement impacts the accuracy of all inferences that rely on the resulting data. While standard sound level meters are well suited for noise studies requiring high precision, their cost, power consumption, and capabilities constrain the scope of application. Alternatively, the wide variety of consumer audio equipment offers many options for acoustical monitoring. The ability to make high resolution, multichannel audio recordings with packages that are relatively small, inexpensive, and low power is especially attractive for long-term acoustical monitoring in remote areas and large-scale spatial surveys that require many devices. These recordings are more valuable when they are calibrated and processed to yield sound level data. Despite the promise of consumer audio equipment, there are several drawbacks and unknowns. Within the framework of the signal chain, this talk will discuss some of the benefits, challenges, and unknowns of using consumer audio equipment for unattended acoustical monitoring.

8:25

3aNS2. Weather focused challenges for continuous monitoring of military noise. Jordan D. Klein, Steven Bunkley, Sahil G. Patel, Richard D. Brown, Jason D. Ray (U.S. Army Engineer Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, jordan.d.klein@usace.army.mil), Jesse M. Barr, Matthew G. Blevins, Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Anton Netchaev (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

Domestic military installations generate high levels of noise due to testing and training which leads to annoyance and complaints from surrounding communities. This necessitates continuous noise monitoring to provide decision makers with the information they need to proactively manage their noise environment. Due to the diverse climates in which military testing and training are conducted (e.g., desert, tundra, and rainforest), monitoring equipment that can operate in a variety of environmental conditions with minimal maintenance and low power consumption is needed. Using existing technologies as a baseline, various iterations of a low-cost acoustic monitor were designed to meet these constraints while minimizing initial investment cost, improving the mean time between failures, and increasing overall system capability. This paper will describe the system developed to provide a rapid deployment option that is robust to extreme temperatures, humidity, and destructive wildlife. A review of operational logs collected during multiple deployments was used to evaluate system performance against benchtop and off-the-shelf solutions. This data demonstrate the reliability of the monitoring stations and the sustainability of their hardware.

8:45

3aNS3. Developments in continuous unattended monitoring systems. Douglas Manvell, Ken Anderson (EMS Bruel & Kjaer, Naerum, Denmark), and Bryce Docker (EMS Bruel & Kjaer, 2330 East Bidwell St., Ste. 210, Folsom, CA 95630, bryce.docker@emsbk.com)

Continuous, unattended noise monitoring systems can immediately alert you should noise levels exceed defined criteria. Once alerted to an exceedance, operators can act to return levels to compliance. This approach has two significant limitations. First, the operator can only take action after the breach has occurred and therefore systems are only able to inform owners about problems that have occurred in the past, rather than allowing them to maintain compliance. Second, the noise limit exceedances might not be due to specific noise from the operator but from unrelated, residual noise in the often-complex noise climates around the particular site and will then be the cause for a false positive. Compliance breaches are frequently triggered by aircraft overflights, road traffic or community sources. Modern monitoring systems enable users to view noise characteristics and listen to the noise breach to determine the source and act if

necessary. However, this approach can create a significant number of false positives each taking up operator time to address. A previous paper by the authors described how airport noise management systems have addressed this problem by combining data from other systems, and how different techniques are required in urban & industrial noise management. This paper describes developments in these techniques and gives examples of techniques that allow operators to take action before a compliance breach occurs, and to reduce the number of false positive alerts.

9:05

3aNS4. Rotorcraft and unmanned aerial system noise measurement technology development and challenges. James H. Stephenson (AMRDC US Army, 2 North Dryden St., MS461, Hampton, VA 23681-2199, james.h.stephenson@nasa.gov), Keith Scudder (AMA, Inc., Hampton, VA), and Eric Greenwood (NASA LaRC, Hampton, VA)

Rotorcraft and Unmanned Aerial Systems (UAS) produce undesired noise that propagates into the community. Significant research has been dedicated over the years to accurately predict the acoustic field of these vehicles from first principles. Despite significant advances in predictive capability, (semi)empirical predictions incorporating acoustic flight test measurements remain the most accurate way to model their noise. However, flight test measurements present significant challenges that must be mitigated. Examples include ground and atmospheric attenuation, inclement weather, and the need to ensure a low background noise condition at the test location. Background noise requirements typically limit testing to remote areas where facility power is not natively available. In order to help mitigate these challenges, a radio-controlled and ground-based measurement system with low space, weight, and power requirements was developed. This system, called Wireless Acoustic Measurement System II (WAMS II), is remotely activated, can record local weather and acoustic measurements with 24-bit accuracy, and has an extended battery life. These capabilities are an improvement on the first generation system and allow for the easy deployment of the multiple systems required to capture the full vehicle acoustic directivity patterns. The design, development, and effectiveness of these systems will be discussed.

9:25

3aNS5. A broadband impulsive signal detection filter for unknown acoustic sources in outdoor environments using non-coplanar microphone arrays. Steven Bunkley (Engineer Res. and Development Ctr., US Army Corps of Engineers, 3909 Halls Ferry Rd., Vicksburg, MS 39180, steven.l.bunkley@usace.army.mil), Michael J. White, Matthew G. Blevins (Engineer Res. and Development Ctr., US Army Corps of Engineers, Champaign, IL), Anton Netchaev, Jordan D. Klein (Engineer Res. and Development Ctr., US Army Corps of Engineers, Vicksburg, MS), Gregory W. Lyons (Engineer Res. and Development Ctr., US Army Corps of Engineers, Champaign, IL), and richard haskins (Engineer Res. and Development Ctr., US Army Corps of Engineers, Vicksburg, MS)

This paper describes a method for non-coplanar microphone arrays that temporally isolates and cleans unknown broadband acoustic impulses for detection, classification, and scene analysis. Possible events are initially identified using a sliding statistical time window. Then the authors posit that most of the false triggers due to environmental noise can be filtered by using generalized cross correlation to phase align the microphone channels and reject implausible velocities. Finally, the phase aligned signals are calibrated and averaged across the microphones. With appropriate hyperparameter tuning, this method appears robust to ambient noise, wind noise and physical interaction. Performance is measured using a simulation and a real historic dataset of over 2 hours of curated acoustic recordings containing 559 gunshots, 120 blasts, and 747 other various weather and non-impulsive events recorded with no prior information under normal operating conditions. Events were found and validated using human listeners with a tool to visualize the waveform and the spectrogram. For this dataset, the model accurately found over 95% of the gunshots with 92% temporal separation and 100% of the blasts identified by the listeners. These results show the method to be a viable solution for impulsive outdoor broadband acoustic signal detection.

Contributed Papers

9:45

3aNS6. Supervised machine learning for crowd noise classification at collegiate basketball games. Kolby T. Nottingham, Katrina Pedersen, Xin Zhao, Brooks A. Butler, Spencer Wadsworth, Blake Smith, Mark K. Transtrum, Kent L. Gee, and Sean Warnick (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT, katrina.pedersen@gmail.com)

Acoustical monitoring combined with machine learning (ML) may help in understanding crowd dynamics. While ML has been applied in numerous audio applications, the aim is usually to distinguish events from noise, rather than trying to characterize the noise itself. This paper comprises an initial study using ML to characterize crowd dynamics during collegiate basketball games. High-fidelity crowd noise recordings from several men's and women's games were synchronized with game video and used to produce a training dataset for supervised ML by linking game events (e.g., baskets, fouls) with acoustic labels (e.g., cheering, silence, and applause). Using the training dataset, a ML classifier was built to identify causal game events from acoustic crowd responses. Findings, potential improvements, and additional crowd noise applications are discussed.

10:00–10:15 Break

10:15

3aNS7. Exploring acoustical approach for pre-screening of building envelope airtightness level. Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca)

Air infiltration plays a significant role in designing and evaluating the performance and air quality of a building. Air leakage through an existing building enclosure can both be localized and calculated by using experimental measurement, such as blower door test system, tracer gas method, and transient approach. Estimating building air permeability through these methods can be expensive, time consuming, and weather reliant. The economical and environmental effect of air infiltration through building envelope leads towards more research on detecting air leakage locations and estimating air infiltration rate through newly introduced techniques, such as using an acoustical method. As such, in this research, a general review of airtightness detection and quantification method will be presented, especially acoustical technique which will be explored more deeply. Moreover, due to significant impact of window systems on total air infiltration through building envelope, the correlation between sound transmission loss and air permeability through seven window assemblies in an existing building will be explored in order to investigate acoustical method further. In addition, the acoustic air leakage detection method based on standard ASTM E1186 will be

instigated on the experimental windows. The results reveal the negative relationship between these two phenomena.

10:30

3aNS8. Counting and tracking vehicles using acoustic vector sensors. Józef Kotus (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland) and Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

A method is presented for counting vehicles and for determining their movement direction by means of acoustic vector sensor application. The assumptions of the method employing spatial distribution of sound intensity determined with the help of an integrated 3D intensity probe are discussed. The intensity probe developed by the authors was used for the experiments. The mode of operation of the algorithm is presented in conjunction with noise characteristics produced by moving vehicles. The optimization of the algorithm is based on measurements of intensity of sound emitted by the vehicle under controlled conditions. A test setup was built for this purpose with the use of measuring devices installed along a road with varying traffic flow. Reference data on the number of vehicles and traffic directions were prepared employing a recorded video and a reference traffic analyzer operating in lidar technology. It is shown that the developed acoustic method may contribute to an increase of effectiveness of commonly used vehicle counting systems employing inductive loops or Doppler radars. [Project financed by the by the Polish National Centre for Research and Development (NCBR) from the European Regional Development Fund under the Operational Programme Innovative Economy No. POIR.04.01.04-00-0089/16 "INZNAK—Intelligent road signs...".]

10:45

3aNS9. Mitigating the pitfalls of testing unmanned aerial system vehicles and components in anechoic chambers. James H. Stephenson (AMRDC US Army, Hampton, VA), Daniel Weitsman (Univ. of Hartford, MS461, Hampton, VA 23681, WEITSMAN@hartford.edu), and Nikolas S. Zawodny (NASA LaRC, Hampton, VA)

Flow recirculation is known to develop inside a closed anechoic chamber when testing unmanned aerial system (UAS) rotor components and vehicles. This flow recirculation modifies the inflow through the vehicle's rotors, which results in significant impacts to the measured acoustic signature. A measurement campaign was undertaken at NASA Langley Research Center in which a UAS rotor was tested inside a small anechoic wind tunnel. Acoustic signatures were obtained with the downwash exhausting down the wind tunnel, and with the tunnel exhaust plugged forcing flow recirculation. Several methods to mitigate the acoustic impacts of flow recirculation were then employed with the wind tunnel in the plugged configuration. The effectiveness of the mitigation strategies are discussed, along with implications for future testing and standards development.

11:00

3aNS10. Military and police small arms firing range noise assessments. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Firing ranges are by their nature sources of very loud noise that can damage the hearing of on-site shooters and observers while also potentially causing a noise disturbance off-site. Small arms firing ranges are typically indoor ranges when located in urban or suburban settings, but when located in rural settings, they are typically outdoor ranges designed with varying types and degrees of noise abatement or no noise abatement at all, depending on the setting. Public firing ranges can be busy and in continuous use under good lighting and favorable weather conditions. Often there are wait times. Military and police (M&P) firing ranges are typically active under scheduled usage times. Both M&P ranges host a variety of small arms, from fully automatic and semi-automatic weapons to bolt-action sniper rifles and large caliber side arms, all of which are very loud. It is often mandatory for M&P personnel to undergo shooting practice on a regular basis with some assigned personnel having certification or qualification performance requirements. Several examples of military and police firing ranges, with their issues, similarities and differences, are discussed. Personnel, interior and/or exterior noise abatement measures are usually necessary for these indoor and outdoor M&P firing ranges.

11:15

3aNS11. Reactive acoustic liner design. Ramani Ramakrishnan (Architectural Sci., Ryerson Univ., 325 Victoria St., Toronto, ON M5B 2K3, Canada, rramakri@ryerson.ca) and David H. Van Every (Aiolos Eng., Toronto, ON, Canada)

Acoustic treatment to reduce fan noise levels in a wind tunnel circuit consists of fibrous materials such as fibreglass or rockwool. Open cell foam materials are also used as acoustic treatments. The acoustic treatments are conventionally applied at fan tail cone regions, tunnel walls along fan diffuser section, cross-legs, test section diffuser, and nozzle contraction areas. However, conventional treatments are not possible in cryogenic wind tunnels, since bulk absorber materials with required resistivity, when operating at cryogenic temperatures, are not available. One possible solution is to design reactive silencers tuned to dominant frequencies. One such approach was used as noise control technique so as to satisfy test section noise specifications. The sound power spectrum of the compressor at different speeds were evaluated. The estimated test section sound pressure levels showed noise reduction at two dominant frequencies were required. The acoustic treatment, therefore, resulted in a double layer reactive design tuned to the two dominant frequencies. The design process will be highlighted in the presentation. The final treatment details will also be presented.

WEDNESDAY MORNING, 7 NOVEMBER 2018 RATTENBURY A/B (FE), 8:30 A.M. TO 11:45 A.M.

Session 3aPA

Physical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Willis Coupling in Acoustic Metamaterials

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Feruza Amirkulova, Cochair
Mechanical Engineering, Western New England University, 1215 Wilbraham Road, Springfield, MA 01119

Invited Papers

8:30

3aPA1. Bianisotropic acoustic metasurfaces for wavefront transformation. Steven Cummer, Junfei Li, Chen Shen (Dept. of Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, cummer@duke.edu), Ana Díaz-Rubio, and Sergei Tretyakov (Dept. of Electronics and NanoEng., Aalto Univ., Aalto, Finland)

Acoustic metasurfaces are thin, engineered structures that can control the local reflection and transmission phase of acoustic waves, and thus enable a high degree of flexibility for wave manipulation. Here, we describe our recent work on so-called perfect metasurfaces, in which elements are designed that can control the local transmission and reflection amplitude and phase response of the metasurface. Controlling both the amplitude and the resulting asymmetric phase response requires a fundamentally asymmetric surface impedance for the unit cells. The resulting ideal surface properties thus have strong links to the concept of bianisotropy in electromagnetics and Willis coupling in elastodynamics. We have developed a shunt resonator-based element design that contains the needed degrees of freedom to independently control the surface impedance on both sides of the metasurface. Using this perfect metasurface design approach, we demonstrate the design and experimentally measurement of high efficiency wide-angle transmissive beam steering at levels beyond what is possible with phase control alone.

8:50

3aPA2. Willis coupling in underwater elastic Helmholtz resonators. Xiaoshi Su and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Helmholtz resonators (HRs) have been central in many acoustic metamaterial devices. For example, one can use an array of HRs to obtain negative effective bulk modulus or a panel of HRs to achieve total absorption of low-frequency sound. The aforementioned applications are based on the local monopolar resonance of each HR. However, the HR behaves differently in water. The elastic modulus of the resonator wall can become comparable to the modulus of water making the resonant frequency much lower than the rigid wall case. Moreover, considering the wall elasticity and mass leads to large structural asymmetry and induces cross-coupling between pressure and velocity fields. In this talk, we provide a lumped element model in order to predict the resonant frequency of the elastic HR. The model is demonstrated by sound scattering from an elastic HR in the context of acoustic bianisotropy, or Willis behavior, following a recent paper [Li Quan *et al*., Phys. Rev. Lett., 2018]. It is found that both the pressure and velocity fields can generate monopole and dipole responses from an elastic HR. The explicit example of an elastic HR in a one-dimensional waveguide will be discussed.

9:10

3aPA3. Dynamic homogenization of spatio-temporal metamaterials. Daniel Torrent (Phys., Universitat Jaume I, Av. de Vicent Sos Baynat, s/n, Castellon de la Plana, Castellón 12071, Spain, dtorrent@uji.es)

We discuss the homogenization of acoustic metamaterials with spatio-temporal modulation. It is shown that the effective medium required for the correct description of these structures is a non-recyprocal Willis material. Analytical expressions are given for the effective parameters, and several numerical examples are shown. The theory is applied as well to the specific case of thermal metamaterials, and it is shown that a thermal diode is feasible by spatio-temporal modulation.

9:30

3aPA4. Wave propagation in 1D piezoelectric crystal coupled with 2D infinite network of capacitors. Anton A. Kutsenko (Appl. Mathematics, Jacobs Univ., Campus Ring 1, Bremen 28759, Germany, a.kutsenko@jacobs-university.de)

For 1D piezoelectric crystal, the dispersion equation is a simple linear dependence between the frequency and the wave number. After connecting the crystal to the 2D electrical network of variable capacitors, we can obtain very unusual and tunable dispersion diagrams. We present analytic results describing dispersion equations for surface and volume acousto-electric waves and show the corresponding wave simulations. The talk is based on the papers: 1) A. A. Kutsenko, A. L. Shuvalov, and O. Poncelet, "Dispersion spectrum of acoustoelectric waves in 1D piezoelectric crystal coupled with 2D infinite network of capacitors," *J. Appl. Phys.* **123**, 044902 (2018); 2) A. A. Kutsenko, A. L. Shuvalov, O. Poncelet, and A. N. Darinskii, "Tunable effective constants of the one-dimensional piezoelectric phononic crystal with internal connected electrodes," *J. Acoust. Soc. Am.* **137**(2), 606–616 (2015).

Contributed Papers

9:50

3aPA5. Acoustic bianisotropic metasurfaces for broadband non-reciprocal sound transport. Bogdan Ioan Popa, Yuxin Zhai, and HyungSuk Kwon (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

Acoustic bianisotropic materials are media in which the unconventional momentum—strain and stress—particle velocity couplings offer rich mechanical wave dynamics. For instance, transformation elastodynamics have revealed that bianisotropic media could enable extraordinary control over the propagation of elastic waves including the ability to cloak regions of space from detection with mechanical waves. The development of metamaterials have provided the tools to implement these unconventional materials and the first experimental acoustic bianisotropic materials have featured interesting properties such as independent engineering of transmission and reflection. However, despite recent work in this area, the wave dynamics enabled by mechanical bianisotropy have not been fully uncovered and explored. In this presentation we demonstrate experimentally the power of active bianisotropic metasurfaces to produce highly non-reciprocal sound transport in free space in a remarkably broadband fashion and in very compact materials. In addition, we show that this unusual mechanism for non-reciprocal sound propagation is truly linear and does not rely on any type of frequency conversion or other non-linear processes.

10:05–10:20 Break

10:20

3aPA6. Acoustic scattering from a radially-polarized Willis-coupled cylinder. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil), Andrew J. Lawrence, Benjamin M. Goldsberry, and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This talk presents analytical and numerical solutions to the scattering of plane waves from a Willis cylinder with radial polarization. Assuming weak coupling, the scattered field may be explicitly written as a power series expansion of the coupling strength. Using an appropriate choice of average mass density, bulk modulus, and Willis coupling, it is found that the first three scattering modes may be suppressed, providing partial acoustic cloaking using a scattering cancellation approach similar the work of Guild *et al.* [*Phys. Rev. B*, **86**, 104302 (2012).]. Furthermore, we find that the Willis coupling does not need to be strong in order to achieve this mode suppression and so the weak-coupling formulation is appropriate. The scattering problem is then solved numerically using finite element analysis using the weak form of the governing equations, which is then compared to the approximate analytical model for validation. [Work supported by U.S. Army ERDC Geospatial Research and Engineering business area and by ONR.]

10:35

3aPA7. Guided waves at bianisotropic fluid interfaces. Samuel P. Wallen (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu), Caleb F. Sieck (NRC Postdoctoral Res. Associate Program, U.S. Naval Res. Lab., Washington, DC), Benjamin M. Goldsberry (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Matthew D. Guild, Gregory Orris (U.S. Naval Res. Lab., Washington, DC), and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Willis fluids are characterized by constitutive relations that couple the pressure and momentum density to both the particle velocity and the volume strain. This effective dynamic response coupling may arise due to microstructural asymmetry, long range order, or time-varying material properties and has been shown to be analogous to electromagnetic bianisotropic media [Phys. Rev. B **96**, 104303 (2017)]. In this study, we report on the existence of guided waves at the interface between two fluids when at least one displays Willis coupling. Criteria for the existence of these waves are discussed in terms of the material properties, frequency, and wave number, and expressions for the dispersion relation and rate of spatial decay away from the interface are obtained analytically. We demonstrate that interface waves are supported when one of the fluids possesses Willis coupling, in contrast to an interface between two classical isotropic fluids, which cannot support interface waves. Special cases are highlighted via numerical examples. [Work supported by ONR, NSF, the Applied Research Laboratories at The University of Texas at Austin, and the NRC Research Associateship Program.]

10:50

3aPA8. Acoustic scattering by a Willis-coupled fluid cylinder. Feruza Amirkulova (Mech. Eng., San Jose State Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza@scarletmail.rutgers.edu) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers The State Univ. of New Jersey, Piscataway, NJ)

We present an integral equation based approach to analyze an acoustic scattering from fluid cylinders exhibiting Willis coupling. The Willis-coupled cylinder is characterized by a bulk modulus and mass density as well as a coupling vector. The coupling vector relates the pressure and momentum density to the volume strain and particle velocity simultaneously. We consider integral representations obtained using the free-space Green's function for the exterior fluid medium. The integral equations are evaluated using the T-matrix method. The computation of scattered pressure wave due to an obliquely incident wave on bi-anisotropic fluid cylinder immersed in an external homogeneous fluid will be described. The method will be illustrated giving examples for fluid Willis cylinders.

11:05

3aPA9. Acoustic bianisotropy in effective fluids of finite domain. Caleb F. Sieck, Matthew D. Guild (Code 7160, NRC Postdoctoral Res. Associateship Program, U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, caleb.sieck.ctr@nrl.navy.mil), Andrea Alù (Dept. of Phys. and Elec. Eng., City Univ. of New York, New York, NY), Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Gregory Orris (Code 7160, U.S. Naval Res. Lab., Washington, DC)

Recent research has shown that subwavelength asymmetry and/or nonlocal effects in heterogeneous acoustic media can be described as a dynamic effective fluid displaying acoustic bianisotropic properties. Theoretical homogenization schemes have demonstrated emergent acoustic bianisotropy for an infinite array of subwavelength inhomogeneities in a fluid matrix, and experimental studies have demonstrated individual asymmetric microstructures and single layer metasurfaces exhibiting bianisotropy. However, metamaterials research has shown that a finite domain effective fluid behaves unlike an infinite domain or a surface, and the effective properties of the domain tend to depend on the chosen boundaries and on the position within the domain, even when the microstructure is periodic. This work presents the use of multiple scattering homogenization to discuss the dependence of bianisotropic properties of finite effective fluids on the choice of boundaries and domain size in order to inform future experimental studies on acoustic bianisotropy. [Work supported by the Office of Naval Research and the NRC Research Associateship Program.]

11:20

3aPA10. Acoustic focusing via guided waves in bianisotropic fluid layers. Samuel P. Wallen (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu), Andrew J. Lawrence, Benjamin M. Goldsberry (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Willis fluids display a complex, effective dynamic response stemming from microstructural asymmetry, long range order, and/or time-varying material properties. These materials are characterized by constitutive relations that couple the pressure and momentum density to both the particle velocity and the volume strain. This coupling has been shown to be analogous to electromagnetic bianisotropic media, which exhibit coupled electric and magnetic fields [Phys. Rev. B **96**, 104303 (2017)]. In a recent study, an acoustic lens composed of an array of Willis fluid layers was demonstrated, where the pressure phase imparted by each array element was tuned by varying the direction of Willis polarization and was determined using finite element analysis. In this work, we report on analytical solutions for guided waves in layers of fluid displaying Willis coupling and derive dispersion relations for layers with rigid and pressure-release boundary conditions. Special cases highlighting the effect of Willis polarization direction are presented via numerical examples. The acoustic lens is revisited with array phases obtained from analytical solutions, eliminating the need for finite element methods. Lenses designed for focusing and steering incident pressure waves are demonstrated using fully analytical models. [Work supported by NSF, ONR, and ARL:UT.]

Session 3aPP

Psychological and Physiological Acoustics: Acoustics Bricolage (Poster Session)

Christian Stilp, Chair
Psychological and Brain Sciences, University of Louisville, 308 Life Sciences Building, Louisville, KY 40292

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aPP1. Rippled spectrum resolution in normal listeners: Estimates at different experimental paradigms. Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru), Olga Milekhina, and Dmitry Nechaev (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Rippled-spectrum (RS) signals are used for evaluation of resolution of spectro-temporal structure of sounds. Measurements of spectrum-pattern resolution imply discrimination between the test and reference signals. Therefore, estimates of rippled-pattern resolution may depend not only on the test signal but also on the reference signals. In the present study, the ripples density resolution was measured in combinations of the test and reference signals: (i) RS with phase reversals vs RS without phase reversion; (ii) RS with phase reversals vs non-rippled spectrum; (iii) RS without phase reversals vs non-rippled spectrum; and (iv) RS without phase reversals vs RS of opposite ripple phase. The spectra were centered at 2 kHz and had ERB of 1 oct and SPL of 70 dB. A three-alternative forced-choice procedure was combined with adaptive stimulation procedure. With rippled reference signals, mean ripple resolution limits were 8.9 ripple/oct (phase-reversals test signal) or 7.7 ripple/oct (constant-phase test signal). With non-rippled reference signal, mean resolution limits were 26.1 ripple/oct (phase-reversals test signal) or 22.2 ripple/oct (constant-phase test signal). Different contribution of excitation-pattern and temporal-processing mechanisms is supposed for measurements with rippled and non-rippled reference signals. [Work supported by Russian Science Foundation Grant 16-15-10046.]

3aPP2. Discrimination of band-limited rippled spectra of various central frequencies in cochlear implant users. Dmitry Nechaev (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, dm.nechaev@yandex.ru), Marina Goykhburg, Vigen Bakhshinyan (National Res. Ctr. for Audiol. and Hearing Rehabilitation, Moscow, Russian Federation), Alexander Supin (Inst. of Ecology and Evolution, Moscow, Russian Federation), and George Tavartkiladze (National Res. Ctr. for Audiol. and Hearing Rehabilitation, Moscow, Russian Federation)

Sound signals with wide-band rippled spectra were used for the frequency resolving power measurements in cochlear implant users. In the present study, resolution of band-limited ripple spectra of various central frequencies and speech discrimination test (Russian language) were compared. The rippled spectra had 2-oct cosine envelope with ripples equally spaced on the logarithmic scale. The central frequencies of the rippled spectra were 1, 2, or 4 kHz. Ripple resolution was measured with a three-alternative forced-choice procedure using a paradigm of discrimination of a test signal with ripple phase reversals from a reference signals with a constant-phase ripples. The average ripple discrimination thresholds were 1.9 ripples/octave (RPO) for 1 kHz; 2.3 for 2 kHz; and 2.3 for 4 kHz. The lowest correlation between ripple discrimination limits and speech discrimination results was about 0.2 at a central frequency of 4 kHz, the highest correlation was 0.6 at 1 kHz. In all the cases, the relation between the ripple spectrum resolution and speech discrimination test had positive trends. Band-limited rippled-spectrum tests allow to reveal informative frequency bands for speech discrimination in cochlear implant users. [Work supported by the Russian Science Foundation, Grant No. 16-15-10046.]

3aPP3. Perceiving phonemes with simulated variable cochlear implant insertion depths. Michael L. Smith (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Box 354875, Seattle, WA 98105, smithm59@uw.edu) and Matthew Winn (Speech Lang. & Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Cochlear implant (CI) patients have difficulty encoding spectral aspects of a speech signal, and are at a severe disadvantage compared to normal hearing listeners in distinguishing consonants that differ by place of articulation (PoA). One reason for this difficulty could be due to a shifting of spectral energy that occurs with a shallow insertion depth of the electrode array. The present study aimed to measure the effects of spectral shifting on phoneme categorization and PoA perception specifically. Speech was vocoded and then shifted by 0, 2, 4, and 6 mm in cochlear space to simulate shallow CI insertion depth in normal hearing listeners. Stimuli included a /b-d/ and /s-ʃ/ continua that differed by PoA, as well as /ra/ and /la/, to create a six-alternative forced choice paradigm. Results show phoneme categorization became less categorical, with a biased perception of /b-d/ and /s-ʃ/ continuum towards /d/ and /s/ respectively, consistent with higher-frequency energy in those consonants. Some participants were better at recalibrating to the spectrally shifted stimuli than others, as their perception of each continua remained somewhat balanced with increased amounts of spectral shifting. This study showcases the potential consequences and individual differences resulting from shallow insertion depth of a CI.

3aPP4. Cochlear wave propagation under acoustical and electrical stimulations. Amir Nankali (ME Dept., Univ. of Michigan, Ann Arbor, MI 48105, nankali@umich.edu)

Input sounds to the mammalian ear produce waves that travel from the base to the apex of the cochlea. These traveling waves stimulate the microstructure of the organ of Corti (OoC) in the center of the cochlea. The interplay between the traveling wave and a distinct nonlinear amplification process boosts the cochlear responses and enables sound processing over a broad frequency and intensity ranges. The *in vivo* electrical and acoustical stimulations have extensively been used to identify the underlying amplification process. In order to interpret the experimental data and propose a precise active mechanism, a comprehensive three-dimensional model of the cochlea is used. Our model predicts that applying an intracochlear excitation (electrical current or mechanical force) on a location along the cochlea generates a dispersive wave propagation in both forward and reverse directions. It is also found that the wave propagation in the cochlea is nonreciprocal

when stimulated by an electrical current applied in the OoC. This effect is attributed to the feed-forward mechanism, mediated by a longitudinal electrical cable in our model. [This work was supported by NIH Grant Nos. DC-004084 and T32DC-00011.]

3aPP5. Alleviated cochlear damage in an inflammation suppressing model. Hongzhe Li, Liana Sargsyan, and Alisa Hetrick (VA Loma Linda Healthcare System, 11201 Benton St., Res. Service, Loma Linda, CA 92357, Hongzhe.Li@va.gov)

Cochlear inflammatory response to various environmental insults, has been increasingly become a topic of interest. As the immune response is associated with both pathology and protection, targeting specific components of the immune response is expected to dissect the relationships between cellular damage and inflammation-associated protection and repair in the cochlea. Duffy antigen receptor for chemokines (DARC) is a member of a group of atypical chemokine receptors, and essential for chemokine-regulated leukocyte/neutrophil trafficking during inflammation. Previous studies have reported that *Darc* deficiency alters chemokine bioavailability and leukocyte homeostasis, leading to significant anti-inflammatory effects in tissues following injury. In this study, we have used *Darc* knockout mice to determine the impact of a deficiency in this gene on cochlear development, as well as function in cochlea subjected to various stresses. We observed that DARC is not required for normal development of cochlear function, as evidenced by typical hearing sensitivity in juvenile *Darc*-KO mice, as compared to wild type *C57BL/6* mice. However, *Darc*-KO mice exhibited improved hearing recovery after intense noise exposure when compared to wild-type. At cochlear locations above frequency range of the energy band of damaging noise, both hair cell survival and ribbon synapse density were improved in *Darc* deficient animals.

3aPP6. Comparison of wideband and clinical acoustic reflex thresholds in patients with normal hearing and sensorineural hearing loss. M. P. Feeney (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Kim Schairer (James H. Quillen VAMC, PO Box 4000, Mountain Home, TN 37684, kim.schairer@va.gov), Douglas H. Keefe, Denis Fitzpatrick (Boys Town National Res. Hospital, Omaha, NE), Daniel Putterman, Angie Garinis, Michael Kurth (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Elizabeth Kolberg, Kara McGregor (James H. Quillen VAMC, Mountain Home, TN), and Ashley Light (East Tennessee State Univ., Johnson City, TN)

Acoustic reflex thresholds (ARTs) obtained with a wideband (WB) probe and an adaptive threshold detection procedure were compared to ARTs using a clinical system. Ipsilateral and contralateral ARTs were elicited in a group of 79 adults with normal hearing (NH) and 51 adults with sensorineural hearing loss (SNHL). ARTs were obtained for both methods using activator tones of 0.5, 1.0, and 2.0 kHz and broadband noise (BBN) with a bandwidth extending to 4.0 kHz for the clinical and 8.0 kHz for the WB ART. Results were similar for ipsilateral and contralateral ARTs. Tonal ARTs with the clinical method were slightly elevated for the SNHL group for all three activator tones, but for the WB method were elevated at 2.0 kHz where the average hearing loss was greatest (47 dB HL). ARTs for BBN were higher for the clinical method than the WB method for both groups, and the difference between groups was around 5 dB for the clinical method but 12 dB for the WB method. This suggests that ARTs with the WB method and BBN activator extending to 8 kHz are a more sensitive indicator of high-frequency SNHL than the clinical method. Individual reflex patterns will also be presented.

3aPP7. Wideband transient otoacoustic emissions in ears with normal hearing and sensorineural hearing loss. Kim Schairer (James H. Quillen VAMC, PO Box 4000, Mountain Home, TN 37684, kim.schairer@va.gov), Douglas H. Keefe, Denis Fitzpatrick (Boys Town National Res. Hospital, Omaha, NE), Daniel Putterman (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Elizabeth Kolberg (James H. Quillen VAMC, Mountain Home, TN), Angie Garinis, Michael Kurth (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Kara McGregor (James H. Quillen VAMC, Mountain Home, TN), Ashley Light (East Tennessee State Univ., Johnson City, TN), and M. P. Feeney (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

Otoacoustic emissions (OAEs) are generated in the cochlea in response to sound and are used clinically to separate ears with normal hearing from sensorineural hearing loss (SNHL). OAEs were elicited at ambient pressure by clicks (CEOAE) and wideband chirps (TEOAE) sweeping from low-to-high frequency with a sweep rate of either 187.6 Hz/ms (short chirps) or 58.2 Hz/ms (long chirps) and a bandwidth extending to 8 kHz. Chirps were presented at the same sound exposure level (SEL) as clicks, or + 6 dB relative to clicks. A total of 288 OAE waveforms were averaged for short chirps in ~1 minute compared to 120 waveforms for long chirps. Compared to clicks, the chirp has a lower crest factor, which allows it to be presented at an overall higher SEL without distortion. OAEs were elicited in 79 adults with normal hearing and 51 adults with mild-to-moderate SNHL. One-sixth octave OAE signal-to-noise ratios from 0.7 to 8.0 kHz were compared across stimulus types and conditions. The area under the receiver operating curve (AUC) was used to assess the accuracy of detecting SNHL. Average AUCs across 1/6th octave frequencies ranged from 0.90 to 0.89 for TEOAEs and were 0.87 for the CEOAE suggesting excellent test performance.

3aPP8. Behavioral methodology demonstrating the development of Tinnitus from chronic non-traumatic noise in mice. Kali Burke, Laurel A. Screven (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Matthew A. Xu-Friedman (Biological Sci., Univ. at Buffalo, SUNY, Buffalo, NY), and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Tinnitus is the perception of a ringing or hissing noise in the absence of an external auditory stimulus. It can develop in one or both ears following acute or chronic noise exposure, head trauma, or as the result of natural aging. Using the Stolzberg *et al.* (2013, *J. Neurosci. Methods*) identification paradigm for rats, we assessed tinnitus in awake and behaving mice in response to long-term, non-traumatic noise exposure. Mice were trained to categorize three stimuli into two categories. Category 1 was 1/8 octave narrow band noise bursts centered around 4, 8, 16, 22.6, and 32 kHz. Category 2 was amplitude modulated white noise (modulated 100% at a rate of 5 kHz) and silent trials. Following this training period, mice lived in 85 dB noise for 23 hours a day and were tested for the remaining 1-hour each day. Most of the mice developed tinnitus, which was indicated by a shift in the categorization of the silence trials from category 2 to category 1. The present experiment demonstrates that chronic, non-traumatic noise exposure can induce tinnitus in mice in a similar way to salicylate injections in rats.

3aPP9. The influence of time of hearing aid use on auditory perception in various acoustic situations. Piotr Szymanski (Training and Development Dept., GEERS Hearing Acoustics, Lodz, Poland), Tomasz Poremski (Audiol. Support Ctr., GEERS Hearing Acoustics , Lodz, Poland), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The assessment of sound perception in hearing aids, especially in the context of benefits that a prosthesis can bring, is a complex issue. The objective parameters of the hearing aids can easily be determined. These parameters, however, do not always have a direct and decisive influence on the subjective assessment of quality of the patient's hearing while using a hearing aid. The paper presents the development of a method for the assessment of auditory perception and the effectiveness of applying hearing aids for hearing-impaired people during a short-term use. The method involves a questionnaire based on the APHAB (Abbreviated Profile of Hearing Aid Benefit) assessment questionnaire, a measure of self-reported auditory disability. The study includes additional criteria, such as measuring the number of hours and days of use of hearing aids, the degree of hearing loss and the patient's experience. A web-based application is developed to enable to carry out such an examination from any computer with access to the network. The research results show that in the first period of use of hearing aids, speech perception improves, especially in noisy environments. The perception of unpleasant sounds also increases, which leads to deterioration of hearing aid acceptance by their users.

3aPP10. Development and calibration of a smartphone application for use in sound mapping. Lawrence L. Feth, Evelyn M. Hoglund, Gus Workman, Jared Williams, Morgan Raney, and Megan Phillips (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, feth.1@osu.edu)

Klyn and Feth (2016) reported preliminary work to use a smartphone application in a citizen-science project designed to map sound levels in Columbus, OH. Before the main project began, we discovered that the sound level measuring applications available for download had shortcomings that made them unsuitable for the proposed work. This presentation describes the development of two smartphone applications, one iOS and one Android, and the calibration procedures developed to document their accuracy and reliability. Following the suggestions of Kardous, *et al.* (2014, 2016), we require that the measurements be conducted using an external microphone. In use, microphone voltage is sampled for 30 seconds and processed to reflect the A-weighting scale so that sound levels are recorded as dBA values. The time and location of each sample is saved with the sound level value and can only be uploaded to the project data base if the device has been previously calibrated. Calibration stores an offset value that can be added to each sample before producing mapped values. Limits on proximity of repeated samples are included to ensure a better distribution of results. The apps are currently available for download from their respective "stores." [Work supported by a grant from the Battelle Engineering, Technology and Human Affairs Endowment.]

3aPP11. The role of pitch and harmonic cancellation when listening to speech in background sounds. Daniel Guest and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, guest121@umn.edu)

Fundamental frequency (F0) differences between competing talkers can aid their perceptual segregation (ΔF0 benefit), but the underlying mechanisms remain poorly understood. One theory of ΔF0 benefit, harmonic cancellation, proposes that the periodicity of the masker can be used to cancel (i.e., filter out) its neural representation. Earlier work suggested that a one-octave ΔF0 provided little benefit, an effect predicted by harmonic cancellation due to the shared periodicity of the masker and target. An alternative explanation is that this effect is simply due to spectral overlap between the target and masker. To assess these competing explanations, speech intelligibility of a monotonized target talker masked by a speech-shaped harmonic complex tone was measured as a function of ΔF0 (0, +3, +12, +15 semitones) and spectral structure of the masker (all harmonics, odd harmonics only). We found that removal of the masker's even harmonics when the target F0 was one octave higher than the masker improved speech reception thresholds by about 6 dB. Because this manipulation eliminated spectral overlap between target and masker components but did not alter their shared periodicity, the finding is consistent with the explanation based on spectral overlap, but not cancellation. [Work supported by NIH R01DC005216 and NSF NRT-UtB1734815 grants.]

3aPP12. Acoustic features in speech for emergency perception. Maori Kobayashi (Japan Adv. Inst. of Sci. and Technol., 1-1, Asahidai, Nomi, Ishikawa 9231292, Japan, maori-k@jaist.ac.jp), Yasuhiro Hamada (Japan Adv. Inst. of Sci. and Technol., Nakano, Japan), and Masato Akagi (Japan Adv. Inst. of Sci. and Technol., Nomi, Ishikawa, Japan)

Previous studies have reported that the acoustic features such as the speech rate, fundamental frequency (F0), amplitude, and voice gender are related to emergency perception in speech. However, the most critical factor influencing the emergency perception in speech remains unknown. In this study, we compared influences of three acoustic features (speech rate, F0, and spectral sequence (amplitude)) to determine the acoustic feature that has the most influence on emergency perception in speech. Prior to conducting our experiments, we selected five speech phrases with different level of perceived emergency among various speech phrase spoken by TV casters during real emergencies. We then created synthesized voices by replacing three acoustic features separately among the selected five voices. In experiment 1, we presented these synthesized voices to 10 participants and asked them to evaluate levels of the perceived emergency of each voice by the magnitude estimation method. The results from experiment 1 showed that F0 was most influential on emergency perception. In experiment 2, we examined influences of the three acoustic features on auditory impression related to the perceived emergency by the SD method. The results suggested that emotional effects of some words such as "tense" or/and "rush" were influenced by the fundamental frequency.

3aPP13. Irrelevant sound effects with locally time-reversed speech: Native vs. non-native language. Kazuo Ueda, Yoshitaka Nakajima (Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ueda@design.kyushu-u.ac.jp), Wolfgang Ellermeier (Technische Universitaete Darmstadt, Darmstadt, Germany), and Florian Kattner (Technische Universitaete Darmstadt, Darnstadt, Germany)

To disentangle the contributions of local and global temporal characteristics of irrelevant speech in native and non-native language, irrelevant speech/sound effects (ISEs) of normal speech, reversed speech, locally time-reversed speech [Ueda *et al.* (2017). Sci. Rep. 7:1782] and its reversal on serial-recall of visually presented series of digits were examined. ISE experiments were performed with German native listeners (n= 79) and with Japanese native listeners (n= 81), employing both German and Japanese speech with either sample. All conditions involving speech significantly impaired memory performance when compared to a pink-noise control condition. When the native language of each group of listeners was presented, locally time-reversed speech with the shortest segment duration (20 ms), which was highly intelligible, was as equally disruptive as normal speech to the memory task, whereas locally time-reversed speech with longer segment durations (70 or 120 ms) was less disruptive. The effect of segment duration totally disappeared when the locally time-reversed speech was played backward. When the non-native language was presented, locally time-reversed speech showed the same pattern of results in each language group, irrespective of the direction of playback. Thus, the irrelevant sound effect worked differently for one's native and non-native languages.

3aPP14. Low-rate frequency modulation detection across the lifespan. John Grose and Emily Buss (Univ. of North Carolina at Chapel Hill, Dept. OHNS, CB#7070, 170 Manning Dr., Chapel Hill, NC 27599, john_grose@med.unc.edu)

Frequency modulation (FM) detection at a 2-Hz rate is thought to rely on temporal cues. For a 500-Hz carrier, this reliance is heightened by the use of dichotic presentation. Here, the modulator is inverted across ears such that binaural beats can cue FM. Detection acuity for FM is better in the dichotic than diotic condition, promoting the task as a measure of temporal fine structure (TFS) sensitivity. Older adults show less benefit of dichotic presentation than young adults, suggesting deficient TFS processing. This study extended the work to school-age children in order to gain a perspective of TFS processing across the lifespan. Children aged 4–10 yrs showed poorer FM detection in both diotic and dichotic conditions re young adults, and their performance was more similar to that of older adults. The dependence of the task on TFS processing was further tested by adding background noise. This reduced FM detection acuity but more for the dichotic than diotic condition. Overall, these results support the dependence of the FM detection task on TFS processing and demonstrate that such processing appears compromised at both ends of the age spectrum. [Work supported by NIDCD DC01507 (JHG) and DC00397 [EB].)

3aPP15. Loudness constancy in healthy older adults: Effects of sound production and visual cues of music playing. Akio Honda (Shizuoka Inst. of Sci. and Technol., 2200-2 Toyosawa, Fukuroi, Shizuoka 437-8555, Japan, akio.honda6@gmail.com), Ayumi Yasukouchi, and Yoichi Sugita (Waseda Univ., Tokyo, Japan)

Loudness constancy refers to the phenomenon by which loudness remains constant in the presence of substantial changes in a physical stimulus caused by varying the sound distance. We investigated loudness constancy in healthy older adults. The degree of loudness constancy was measured using two methods of adjustment: "sound production," by which listeners played a musical instrument as loudly as a model player, and "sound level adjustment," by which listeners adjusted the loudness of the sound produced by a loudspeaker. The target sound was produced by the actual musical instrument performance. Sound pressure levels of the stimuli were approximately 60, 75, and 86 dB(A). The distances between the performer and the participant were 2, 8, and 32 m. In both conditions, participants were asked to produce the level of sound pressure matching the stimulus. Results show that when visual cues of musical performance are available, sound production had more robust loudness constancy than the sound level adjustment method. These results support an earlier claim that audiovisual perception and imitation are necessary for musical learning and skill acquisition.

3aPP16. Bilingual speech associations in phonetic representation. Margaret Cychosz (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, mcychosz@berkeley.edu) and Erik C. Tracy (Psych., Univ. of North Carolina at Pembroke, Pembroke, NC)

Listeners unconsciously index race, gender, and sexual orientation from little phonetic input. Psycholinguistic theory attributes this to a representation of language where episodic traces encode fine, speaker-specific acoustic detail. We examined this unconscious linguistic bias in a task with native English speaker participants. Participants were presented with two pictures, both of which were either Hispanic or Caucasian males. Simultaneously, they listened to a semantically-neutral English sentence spoken by either a bilingual Spanish-English male or a monolingual English male and were told to choose which pictured man said the sentence. When the auditory stimulus was bilingual speech, speakers were quicker to choose between the two Hispanic faces than between the Caucasian faces. However, when participants heard a monolingual voice, there was no difference in reaction time. In the second phase of the experiment, participants were instead presented with two different pictures, one Hispanic and one Caucasian. Listeners were more likely to associate bilingual English with the Hispanic male than the Caucasian male and vice versa (monolingual English with Caucasian male). These results suggest that listeners quickly index male Spanish-English speech, but monolingual English-Caucasian associations may not be as robust.

3aPP17. Notches in sentence spectra bias subsequent phoneme categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Spectral differences across earlier (context) and later (target) sounds are perceptually magnified, resulting in spectral contrast effects (SCEs) that bias categorization of later sounds. Most studies added spectral peaks to context sounds in order to produce SCEs, but noise contexts with spectral notches at speech frequencies biased phoneme categorization in a complementary manner (Coady *et al*., 2003 *JASA*). We tested whether this approach generalized to speech contexts with spectral notches. On each trial, a context sentence preceded the target phoneme (Experiment 1: /ɪ/-/ɛ/; Experiment 2: /d/-/g/). Sentences were processed by notch filters that attenuated energy in the same frequency regions that, when amplified to add spectral peaks, produced SCEs (Experiment 1: 100–400/550–850 Hz, as in Stilp *et al*., 2015 *JASA*; Experiment 2: 1700–2700/2700–3700 Hz, as in Stilp & Assgari, 2017 *JASA*). Notch depths ranged from −5 to −20 dB in 5-dB steps. In both experiments, notch-filtered sentences biased phoneme categorization in complementary directions to SCEs, with bias magnitudes increasing at larger notch depths. Whether earlier sounds have spectral peaks or notches, speech categorization is highly sensitive to the magnitudes of spectral differences across context and target sounds.

3aPP18. Quasi-phonemic vowels in Ampenan Sasak: An acoustic analysis. Leah Pappas (Linguist, Univ. of Hawai'i at Mānoa, 1890 East-West Rd., Moore Hall, Honolulu, HI 96822, lpappas@hawaii.edu)

Phonological studies on Sasak tend to agree that Sasak has a six-vowel system /i, u, e, o, ə, a/ (Archangeli *et al*., n.d.; Chahal, 1998; Jacq, 1998). However, in Ampenan Sasak, a dialect spoken in the provincial capital of Mataram, it is difficult to attribute vowel quality differences of tense mid-vowels [e, o] and lax mid-vowels [ɛ, ɔ] to allophonic variation of underlying phonemes /e, o/. While lax mid-vowels tend to appear in heavy syllables and tense mid-vowels appear in light syllables (see also Archangeli *et al*., n.d.), there are exceptions (e.g., /baɾeh/ "late," /əmbong/ "reservoir") and several minimal pairs (e.g., /bəɾəmbok̚/ "discuss," /bəɾəmbɔk̚/ "breathe;" /koboʔ/ "levening," /kɔbɔʔ/ "play with water"). Further, speakers' intuitions about these distinctions are unreliable. In order to understand this relationship, we elicited wordlists across 13 female speakers of Ampenan Sasak aged 18 to 48. Wordlists were balanced for syllable weight, coda segment, and syllable position and subjected to acoustic analysis. Preliminary results show that the quality of the vowel is highly affected by the weight of the syllable and the coda segment. This suggests that mid-vowels are quasi-phonemic in Ampenan Sasak (Ladd 2013). While they generally behave as allophones, this paradigm is not strictly followed.

3aPP19. Effects of incomplete feedback on response bias in auditory detection reanalyzed. Shuang Liu (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, Columbus, OH 43210, liu.3267@osu.edu), Matthew Davis (CDO Technologies, Inc., Columbus, OH), and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Davis (2015) reported the effects of providing incomplete feedback to listeners in a simple detection task. When feedback is limited in a detection experiment, the observer's response criterion may deviate significantly from the optimal criterion. To approximate real-world listening conditions, a single-interval yes-no procedure was conducted with 10 feedback conditions ranging from no feedback to complete feedback. The signal was a brief 1 kHz tone; the masker was wideband white noise. The SIAM procedure (Kaernbach, 1990) was used to establish the SNR for 75% detection threshold for each listener. That level was then used in each of the feedback conditions. Davis reported a detailed descriptive analysis of the symmetry, organization, implicitness, and amount of feedback and the individual differences noted across the ten listeners. For this project, an equal-variance Gaussian signal detection framework was used to analyze the data. Model parameters were estimated via Bayesian inference. The main finding is that, as expected, complete feedback drives response criteria toward the optimum, and deviation from the optimal criterion increases as the amount of feedback decreases. While most subjects show this general trend, a few subjects maintain near-optimal behavior throughout all conditions, which is not a surprise.

3aPP20. The effect of experience and tonal contour: A preliminary study of perceived duration on Mandarin tones. Yu-an Lu, Yu-Ming Chang (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, F319, Humanities Bldg. 2, 1001 University Rd., Hsinchu 30010, Taiwan, yuanlu@nctu.edu.tw), Yang-Yu Chen, Peichun Chen, Shao-Jie Jin, Cheng-Huan Lee, Waan-Rur Lu, Yen-Ju Lu, and Yu Nan (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, Hsinchu, Taiwan, Taiwan)

Mandarin tones are shown to be produced with different lengths (i.e., from longer to shorter: T3 > T2 > T1 > T4) (cf. Wu & Kenstowicz, 2015).

An AX rating experiment in which Taiwan Mandarin listeners were asked to rate the relative durations of syllables ([pa], [pi], [ta], [ti]) manipulated into five different duration steps (290 ms, 320 ms, 350 ms, 380 ms, and 410 ms) in Mandarin tones (high-level T1, rising T2, dipping T3, reduced low-level T3, and falling T4) compared with an anchor stimulus ([pa] with 350 ms in mid-level tone) showed that the complex contour tone (T3) was rated as longer than simple contour tones (T2 and T4) and simple contour tones were rated as longer than level tones (T1 and reduced T3). Between the simple contour tones, T2 was rated as longer than T4. Between the level tones, the reduced T3 was rated as longer than T1. The explanations to these tonal perceptual differences are tied to the typological correlation between rime duration and the complexity of tonal targets (T3 > T2/T4 > T1/T3) (e.g., Zhang 2001) as well as to the listeners' experience to the durations of different tones (T2 > T4 and reduced T3 > T1).

3aPP21. Acoustic and articulatory evidence on incomplete syllable-final nasal mergers in Taiwan Mandarin. Chenhao Chiu (Graduate Inst. of Linguist, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenhaochiu@ntu.edu.tw), Yu-an Lu (Foreign Lang. and Literatures, National Chiao Tung Univ., Hsinchu, Taiwan), and Yining Weng (Graduate Inst. of Linguist, National Taiwan Univ., Taipei, Taiwan)

Previous studies reported that syllable-final nasals in Taiwan Mandarin, /n/ and /ŋ/, exhibit merging to different degrees in the vowel contexts of /i/ and /ə/ (e.g., Ing 1985, Kubler 1985, Hsu & Tse 2007, Fon *et al*., 2011). However, the discussion has been mainly on the merging directions, ethnic groups that display the mergers, and attitudes towards the different merging directions. Apart from the large evidence of merging from perceptual identification of the final nasal, this study employs ultrasound imaging technique to examine tongue postures of the syllable-final nasal following /i, a, ə/ vowel contexts. Preliminary results showed that /n/ and /ŋ/ were distinct both acoustically and articulatorily in the context of /a/, echoing earlier findings (e.g., Fon *et al*., 2011). On the other hand, while the contours of tongue shapes demonstrate extensive overlapping between /n/ and /ŋ/ following /i/ and /ə/, their correspondent acoustics display different degrees of nasalization on the preceding vowels (cf. Chen, 2000). More generally, the results suggest that the merging in syllable-final nasals undergoes an incomplete process and the acoustic contrasts might have lie in the nasalization of the preceding vowels and less so in syllable-final nasals.

3aPP22. Effects of corrective feedback on receptive skills in non-native contrasts: A training study in Taiwan Southern Min. YinChing Chang (Graduate Inst. of Linguist, National Taiwan Univ., 4F.-E, No.64-1, Sec. 3, Xinsheng S. Rd., Da'an Dist., Taipei City 106, Taipei 10660, Taiwan, ching199541@gmail.com)

The benefits of corrective feedback (CF) have been proposed in numerous L2 speech perception training studies, concerning that it provides learners with opportunities to retrieve, restructure, and consolidate their phonological representations. Moreover, different types of CF have been investigated considering that different effects will be triggered owing to involving different types of cognitive processes. This study examines the role of different types of CF in receptive skill, focusing on voiceless-voiced contrasts in Taiwan Southern Min, which is absent in Mandarin. Native speakers of Taiwan Mandarin who are also non-native listeners of Taiwan Southern Min are randomly assigned to three groups with different types of CF: Target, Non-target, and Combination of target and non-target. Preliminary results showed that among the three CF groups, identification accuracy significantly improved when participants are provided with combination CF. More generally, the results suggest that corrective feedback can play an influential role in identifying phonemic contrasts that may not occur in listeners' native language.

3aPP23. Coordinated systems in Taiwan Mandarin tone production: An investigation of laryngeal and tongue movements. Chenhao Chiu (Graduate Inst. of Linguist, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Graduate Inst. of Linguist, Taipei 10617, Taiwan, chenhaochiu@ntu.edu.tw)

While electroglottography (EGG) is commonly used to examine different phonation types as well as laryngeal closure and opening of segments, only limited EGG studies focus on tonal quality [Brunelle *et al*., 2010, *Phonetica*, 67:147]. In particular, it is not yet clear how laryngeal movements in terms of its vibration magnitude and vertical displacement may be associated with pitch contour in tone production. The current study tackles this question by examining the amplitude of EGG pulses and laryngeal heights across four tones in Taiwan Mandarin. Ultrasound images are also obtained to assess the temporal relationship between tongue positioning and laryngeal movements during tone production. Preliminary results show that while the laryngeal area is more constricted by low, back vowels, a stronger correlation between EGG amplitude and pitch contour is found. On the other hand, when the tongue is positioned in the front of the cavity, larger degrees of freedom around the laryngeal area yield a lesser correlation between EGG amplitude and pitch contour but a stronger correlation between laryngeal height and pitch contour. The results suggest that tone production may be a result of a combination of laryngeal vibration and spatial coordination between the laryngeal and supralaryngeal movements. [Funding from MOST.]

3aPP24. Frequency-following responses elicited by a consonant-vowel with intonation. Kristin M. Stump and Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu)

To date, two major types of acoustic stimuli have been used to evoke frequency-following responses (FFRs)—either by using a consonant-vowel or by using a vowel with intonation, but not both. The goals of this study were to (1) determine the feasibility of recording FFRs by forging a CV and intonation into one acoustic stimulus, and (2) examine the characteristics of such responses. Twelve Chinese adults (8 females, 4 males, age = 25.58 ± 4.23 years old) were recruited. FFRs were elicited by using the consonant-vowel /da/ combined with a rising intonation pattern (fundamental frequencies 85-93 Hz) for a total of 8000 sweeps from each participant. Six objective indices (consonant amplitude, consonant latency, consonant latency, frequency error, tracking accuracy, and pitch strength) were derived from all recordings. Results demonstrated that it was feasible to record the consonant, vowel, and intonation responses simultaneously. Results also demonstrated distinctive FFR trends with increasing number of sweeps for the consonant, vowel, and intonation responses. Taken together, these findings may have important implications for basic research and clinical applications.

3aPP25. Machine learning in detecting frequency-following responses. Breanna Hart and Fuh-Cherng Jeng (Communications Sci. and Disord., Ohio Univ., 366 Richland Ave., Apt. 2303, Athens, OH 45701, bh130015@ohio.edu)

To improve the efficiency and timeliness in frequency-following response (FFR) testing, the purpose of this study was to investigate the capabilities of machine learning in the detection of an FFR. Continuous brain waves were recorded from 25 Chinese adults in response to a pre-recorded Mandarin monosyllable \yi^2\ with a rising frequency contour. A total of 8000 artifact-free sweeps were recorded from each participant. Continuous

brain waves accumulated (from the first sweep) up to the first 500 sweeps were considered FFR absent; brain waves accumulated (from the first sweep) up to the last 1000 sweeps (i.e., from 7001 to 8000 sweeps) were considered FFR present. Six response features (frequency error, slope error, tracking accuracy, spectral amplitude, pitch strength, and root-mean-square amplitude) were extracted from each recording and served as key predictors in the identification of a response. Twenty-three supervised machine-learning algorithms, with a 10-fold cross-validation procedure, were implemented via a Classification Learner App in MATLAB. Two algorithms yielded 100% efficiency (i.e., 100% sensitivity and 100% specificity) and 14 others produced efficiency $\geq$99%. Results indicated that a majority of the machine-learning algorithms provided efficient and accurate predictions in whether an FFR was present or absent in a recording.

3aPP26. Effects of auditory selective attention on word intelligibility and detection threshold of narrow-band noise. Ryo Teraoka, Shuichi Sakamoto, Zhenglie Cui, Yôiti Suzuki, and Satoshi Shioiri (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, terar@dc.tohoku.ac.jp)

By directing auditory selective attention, humans can recognize a target sound from multiple environmental sounds. Evidence suggests that the effect of auditory selective attention is observed not only in the frequency domain, but also in the spatial domain. However, the effect of auditory spatial attention is not always observed and has been suggested to depend on the level of auditory processing. Therefore, the task dependency of auditory spatial attention was investigated by measuring the word intelligibility and detection threshold of target 1/12 octave-band noise amongst those with different center frequencies. Target and distractor sounds were presented by a loudspeaker array surrounding listeners. The target sound was presented after listeners' auditory attention was attracted to the specific direction from which the target sounds would be presented. Results showed that the word intelligibility increased when attended but the detection threshold of narrow-band noises did not change significantly. This suggests that auditory processing higher than that required for detecting a specific random noise among spatially distributed similar noises is necessary for the effect auditory spatial attention to appear. It seems interesting to pursue the lowest level, where the effects of auditory spatial attention are observable.

3aPP27. Hemispheric decoupling of awareness-related activity in human auditory cortex under informational masking and divided attention. Andrew R. Dykstra (The Brain and Mind Inst., Univ. of Western ON, 1151 Richmond St., Western Interdisciplinary Res. Bldg., London, ON N6A 3K7, Canada, andrew.r.dykstra@gmail.com) and Alexander Gutschalk (Dept. of Neurology, Ruprecht-Karls-Universität Heidelberg, Heidelberg, Germany)

Whether unattended sound streams reach perceptual awareness, and the extent to which they are represented in the central auditory system, are fundamental questions of modern hearing science. We examined these questions utilizing M/EEG, multi-tone masking, and a dual-task dichotic listening paradigm. Listeners performed a demanding primary task in one ear—detecting isochronous target-tone streams embedded in random multi-tone backgrounds and counting within-target-stream deviants—and retrospectively reported their awareness of similar masker-embedded targets in the other ear. Irrespective of attention or stimulation ear, left-AC activity strongly covaried with target-stream detection starting as early as 50 ms post-stimulus. In contrast, right-AC activity, while highly sensitive to stimulation ear, was unmodulated by detection until later, and then only weakly. Thus, under certain conditions, human ACs can functionally decouple, such that one—here, right—is automatic and stimulus-driven while the other—here, left—supports perceptual and/or task demands, including basic perceptual awareness of nonverbal sounds both within and outside the focus of top-down selective attention.

3aPP28. Neural correlates of beat perception in vibro-tactile modalities. Sean A. Gilmore, Gabriel Nespoli, and Frank A. Russo (Psych., Ryerson Univ., 1168 Dufferin St., Toronto, ON M6H 4B8, Canada, sean.gilmore@ryerson.ca)

Musical rhythms elicits a perception of a beat (or pulse) which in turn elicit spontaneous motor synchronization (Repp & Su, 2013). Electroencephalography (EEG) research has shown that endogenous neural oscillations dynamically entrain to beat frequencies of musical rhythms providing a neurological marker for beat perception(Nozaradan, Peretz, Missal, & Mouraux, 2011). Rhythms however, vary on in complexity modulating ability to synchronize motor movement. Although musical rhythms are assumed to be from auditory sources, recent research suggests that rhythms presented through vibro-tactile stimulation of the spine elicits a comparable capacity as auditory in motor synchronization in simpler rhythms, however, this trend diminishes as complexity increases(Ammirante, Patel, & Russo, 2016). The current research purposes to explore the neural correlates of vibro-tactile beat perception with the aim in providing further evidence for rhythm perception from a vibro-tactile modality. Participants will be passively exposed to simple and complex rhythms from auditory, vibro-tactile, and multi-modal sources. Synchronization ability as well as EEG recording will be obtained in order to provide behavioural and neurological indexes of beat perception. Results from this research will provide evidence for non-auditory, vibro-tactile capabilities of music perception.

3aPP29. Analysis of spectral and transmission characteristics of bone-conducted speech using real utterances and transcutaneous vibration. Teruki Toya (School of Adv. Sci. and Technol., Japan Adv. Inst. of Sci. and Technol., 1-1, Asahidai, Nomi-shi 923-1292, Japan, yattin_yatson@jaist.ac.jp), Peter Birkholz (Juniorprofessur für Kognitive Systeme, Technische Universität Dresden, Dresden, Sachsen, Germany), and Masashi Unoki (School of Adv. Sci. and Technol., Japan Adv. Inst. of Sci. and Technol., Nomi-shi, Japan)

Previous studies related to auditory feedback have found that hearing one's own voice has important roles in maintaining one's speech production. However, it is still unclear how perception of one's bone-conducted (BC) speech affects one's speech production, since most studies have just presented white/pink noise for masking speakers' BC speech without enough consideration of acoustical and transmission characteristics of BC speech itself. This study aims to investigate spectral characteristics of the observable parts of BC speech (temporal bone (TB) vibration and ear canal (EC) speech) and transmission characteristics related to them. Spectral envelopes of vowel utterances and long-term average spectrum (LTAS) of sentence utterances were analyzed. Moreover, transfer functions from larynxes to TB and EC were measured using transcutaneous vibration signals. As results of these analyses, although the spectral shapes are different among the vowels in TB vibration and EC speech, it was found that the transmission from larynxes to TB emphasizes lower frequency components and the transmission from larynxes to EC emphasizes the higher frequency components. [Work supported by JSPS KAKENHI Grant No. JP. 17J03679, and Grant in Aid for Scientific Research Innovative Areas (Grant No. 16H01669, 18H05004) from MEXT, Japan.]

3aPP30. Cyclic movement primitives underlying two-handed alternating signs in signed language. Oksana Tkachman, Grace Purnomo, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, tox.cs84@gmail.com)

Biomechanical constraints have been described as underlying patterns in speech production: e.g., cyclic neural central pattern generators (CPGs) in the jaw that evolved for chewing have been claimed to affect syllabic patterns [MacNeilage 1998, *Behav. Brain Sci.* 21: 4] and cyclic patterns in tongue tip movement may span sequences of segments [Derrick & al.

2015, *JASA-EL* 137:3]. Similarly, we propose that otherwise unexplained universal aspects of sign languages may result from a preference for repeated alternating arm movements developed in human ancestors for quadrupedal locomotion. For example, in ASL-LEX corpus of 1000 ASL signs, 159 are balanced (not including signs without movement or articulated in the transverse plane, which has alternative properties we will address in the talk). Of those, 60% are alternating, and the alternating group had fewer iconic signs (30.5% vs 41%) and more repeated signs (61% vs 30%) than symmetrical signs. While both groups had a large proportion of signs realized on coronal plane, which makes them more visible in face-to-face communication (33% for alternating vs 52% for symmetrical), the former have more signs in midsagittal and coronal/transverse planes (63% vs 44%), as predicted. Supporting data from additional sign languages will be presented.

WEDNESDAY MORNING, 7 NOVEMBER 2018 SALON A (VCC), 8:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication, Musical Acoustics, and Psychological and Physiological Acoustics: The Sound of Emotion

Shae D. Morgan, Cochair

Communication Sciences and Disorders, University of Utah, 390 South 1530 East, Suite 1201, Salt Lake City, UT 84112

Kathrin Rothermich, Cochair

Communication Sciences and Disorders, East Carolina University, 355 Beasley Drive, F3, Greenville, NC 27834

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. The acoustic and neurobiological bases of the processing of vocal emotion. Sophie K. Scott (Inst. for Cognit. Neurosci., UCL, 17 Queen Square, London WC1N 3AZ, United Kingdom, sophie.scott@ucl.ac.uk)

In this talk, I will briefly address the ways that the expression of emotion in the voice is affected by physiological processes (e.g., the effects of adrenaline on the vocal folds), as well an interplay of voluntary and involuntary aspects of vocal control. I will outline some behavioural and functional imaging studies of the perception of emotional voices, and the ways that emotion in the voice can be associated with some general acoustic properties (e.g., duration correlates negatively with ratings of arousal). However there is weaker evidence for a relationship between emotional valence and acoustic factors, as well as considerable evidence for emotion specific relationships between acoustic profiles and emotion ratings. I will extend this into a consideration of the sensori-motor networks recruited by different kinds of emotional vocalisations, and argue that this can often reflect more emotion specific effects, which can be linked to both affective and behavioural aspects of emotional processing.

8:25

3aSC2. The experience of music emotion in older adults with hearing loss and hearing aids. Frank A. Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M4L 3T4, Canada, russo@ryerson.ca) and Emma Scholey (Psych., Univ. of Surrey, Guildford, United Kingdom)

The current study examined the effect of hearing loss and hearing aids in older adults on the experience of music emotion. Fifty-four participants aged 60 and above were recruited, including 18 with normal hearing (NH), 18 with hearing impairment (HI), and 18 with hearing impairment who use hearing aids (HA). Arousal ratings and skin conductance responses were obtained from participants across 24 extracts of film music that were previously validated as conveying one of four emotions: happy, sad, fearful, and tender. These four emotions represent a crossing of arousal (high and low) and valence (positive and negative). For each group, an "arousal range" was calculated as the average arousal ratings of happy and fearful excerpts, minus the average arousal ratings of tender and sad excerpts. A similar scheme was used for each group to determine a "skin conductance range". For negatively-valenced music, groups did not differ with regard to arousal of skin conductance. For positively valenced music, NH and HA yielded a larger arousal range than did HI. A similar pattern emerged from the analysis of skin conductance range. Results will be discussed with regard to the acoustic cues to emotion in music and signal processing methods in hearing aids.

8:45

3aSC3. Prosody as a window into speaker attitudes and interpersonal stance. Marc D. Pell, Nikos Vergis, Jonathan Caballero, Maël Mauchand, and Xiaoming Jiang (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montreal, QC H3A1G1, Canada, marc.pell@mcgill.ca)

New research is exploring ways that prosody fulfils different social-pragmatic functions in spoken language by revealing the mental or affective state of the speaker, thereby contributing to an understanding of speaker's meaning. Prosody is often pivotal in signaling speaker attitudes or stance in the interpersonal context of the speaker-hearer; in contrast to the vocal communication of emotions, the significance of prosodic cues serving an emotive or interpersonal function is often dependent on the type of speech act being made and other contextual-situational parameters. Here, we discuss recent acoustic-perceptual studies in our lab demonstrating how prosody marks the interpersonal stance of a speaker and how this information is used by listeners to uncover social intentions that may be non-literal or covert. Focusing on how prosody operates in the communication of politeness, ironic attitudes, and sincerity, our data add to research on the acoustic-perceptual characteristics of prosody in these interpersonal contexts. Future implications for how acoustic cues underlying "prosodic attitudes" affect the interpretive process during on-line speech processing and how they influence behavioral outcomes are considered.

9:05

3aSC4. Acoustics and emotion in tonal and non-tonal languages: Findings from individuals with typical hearing and with cochlear implants. William F. Katz, Cecilia L. Pak (Callier Ctr. for Commun. Sci. and Disord., The Univ. of Texas at Dallas, 1966 Inwood Rd., TX 75235, wkatz@utdallas.edu), and Sujin Shin (Callier Ctr. for Commun. Sci. and Disord., The Univ. of Texas at Dallas, Richardson, TX)

Talkers of tonal language, such as Mandarin, use the acoustic cues of fundamental frequency (F_0), amplitude, and duration to indicate lexical meaning as well as to express linguistic and emotional prosody. It has therefore been hypothesized that tonal language talkers have less prosodic "space" to signal emotional prosody using F_0, compared to non-tonal language talkers, and that these differences in prosodic processing should be evident in speech perception and production tasks. In addition, for talkers of both tonal and non-tonal languages, speaking rate interacts with emotional expression, with "sad" mood generally expressed with slower speech, while "happy" and "angry" moods are marked with faster speech rates. Despite the overall importance of speaking rate in signaling particular emotional moods, few data exist for speakers of tonal languages, such as Mandarin, in which lexical tones are typically specified for length. These issues can be addressed by analyzing and modelling data for individuals with cochlear implants, electronic hearing systems that provide relatively good temporal resolution but poor spectral resolution. Findings from our laboratory will be used to address models of prosody perception and production.

9:25

3aSC5. How do babies laugh? Disa Sauter (Dept. of Psych., Univ. of Amsterdam, Niewe Achtergracht 129B, Amsterdam 1018 XA, Netherlands, d.a.sauter@uva.nl), Bronwen Evans (Univ. College London, London, United Kingdom), Dianne Venneker, and Mariska Kret (Leiden Univ., Leiden, Netherlands)

Laughter occurs across all great ape species, yet human laughter differs from that of other primates: Human laughter is primarily produced on the exhale, whereas other primates laugh on both the inhale and exhale. In the current study, we asked whether human infants laugh in a similar manner to apes, given that human infants, like non-human primates, tend to laugh in the context of tickling or rough-and-tumble play. Human adults, in contrast, laugh across many different kinds of social interactions. To test this hypothesis, we examined whether human infant laughter is acoustically more similar to non-human apes' laughter. Laughter clips from infants aged 3 to 18 months were annotated by phoneticians and evaluated by two listener samples (naïve listeners and phoneticians, respectively). The results provide support for the prediction that the proportion of infants' laughter produced on the exhale increases with age. These results suggest that at younger ages, human infants' laughter is more similar to that of other great apes. These findings are discussed in the context of vocal control maturation and social learning.

9:45–10:00 Break

10:00

3aSC6. Perception and production of vocal emotions by listeners with normal hearing and with cochlear implants. Monita Chatterjee, Jenni Sis, Sara Damm, and Aditya M. Kulkarni (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, monita.chatterjee@boystown.org)

Informative cues for vocal emotions in speech include voice characteristics, speaking rate and intensity. For listeners with cochlear implants, voice characteristics such as pitch and vocal tract length are weakly represented in the electric signal; intensity cues are weakly to moderately preserved, and speaking rate is well represented. Our studies indicate that school-age children (6–18 years of age) with cochlear implants have significant deficits in their recognition of vocal emotions relative to normally hearing peers, similar to post-lingually deaf adults with cochlear implants. However, in contrast to the post-lingually deaf adults, children with cochlear implants showed large deficits in their *productions* of vocal emotions (happy/sad contrasts), measured in 1) acoustic analyses of the produced speech and 2) normally hearing listeners' perceptions of the produced emotions. In contrast to emotions, the words and sentences produced by the children with cochlear implants were highly intelligible. Post-lingually deaf adults with cochlear implants, however, showed excellent performance in the voice emotion production task. These results suggest that the perception of vocal emotions by listeners with cochlear implants may be limited primarily by device limitations, while emotion productions in speech are shaped more strongly by early auditory experience.

10:20

3aSC7. Acoustic cues for vocal emotion recognition by normal-hearing listeners and cochlear implant users. Xin Luo and Kathryn Pulling (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Tempe, AZ 85287, xinluo@asu.edu)

Our recent study found that cochlear implant (CI) users' quality of life in auditory, psychological, and social functioning were predicted by vocal emotion rather than sentence recognition scores. To eventually improve vocal emotion recognition with CIs, this study investigated the acoustic cues for vocal emotion recognition by CI users with CI alone or bimodal fitting, as compared to normal-hearing (NH) listeners. Sentence duration, mean fundamental frequency (F0), and F0 range were individually normalized for emotional utterances of each talker and sentence. In two other conditions, emotional utterances were presented backward in time and with upside-down F0 contours, respectively. Perceptual results showed significant effects of subject group, cue condition, talker, and emotion. Time-reversed utterances worsened NH listeners' recognition of all emotions except sad, while upside-down F0 contours worsened that of angry and happy. Vocal emotion recognition with CI alone only degraded with time-reversed utterances. Time-reversed utterances worsened bimodal CI users' recognition of angry and neutral, while upside-down F0 contours worsened that of angry and happy. Bimodal CI users and NH listeners were also affected by mean F0 and F0 range normalization when recognizing happy. We conclude that natural F0 contours should be faithfully encoded with CIs for better vocal emotion recognition.

10:40

3aSC8. Decoding musical and vocal emotions. Sebastien Paquette (Neurology, BIDMC - Harvard Med. School, 330 Brookline Ave., Palmer 127, Boston, MA 02215, spaquet1@bidmc.harvard.edu)

Many studies support the idea of common neural substrates for the perception of vocal and musical emotions. It is proposed that music, in order to make us perceive emotions, recruits the emotional circuits that evolved mainly for the processing of biologically important vocalizations (e.g., cries, screams). Although some studies have found great similarities between voice and music in terms of acoustic cues (emotional expression) and neural correlates (emotional processing), some studies reported differences specific to each medium. However, it is possible that the differences described may not be specific to the medium, but may instead be specific to the stimuli used (e.g., complexity, length). To understand how these vocal and musical emotions are perceived and how they can be affected by hearing impairments, we assessed recognition of the most basic forms of auditory emotion (musical/vocal bursts) through a series of studies in normal hearing individuals and in cochlear implant users. Multi-voxel pattern analyses of fMRI images provide evidence for a shared neural code for processing musical and vocal emotions. Correlational analyses of emotional ratings helped highlight the importance of timbral acoustic cues (brightness, energy, and roughness) common to voice and music for emotion perception in cochlear implant users.

11:00

3aSC9. Comparing theory, consensus, and perception to the acoustics of emotional speech. Peter M. Moriarty (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, moriarty@psu.edu), Michelle Vigeant (Graduate Program in Acoust., The Penn State Univ., State College, PA), Pan Liu (Univ. of Western ON, State College, PA), Rick Gilmore, and Pamela Cole (Dept. of Psych., The Penn State Univ., State College, PA)

A new corpus of emotional speech was created to conduct theory and evidence-based comparisons to the published literature on the effects of a speaker's intended emotion on the acoustics of their speech. Fifty-five adults were recorded speaking scripts with happy, angry, sad, and non-emotional prosodies. Variations in acoustic parameters such as pitch, timing, and formant deviations were investigated. Based on Scherer's (1986) theoretical predictions about differences between discrete emotions, and Juslin and Laukka's (2003) empirically-derived meta-analytic conclusions, we measured the degree to which the emotional speech data reflected predicted differences in the acoustic parameters of these prosodies. First, in relation to non-emotional prosody, angry and happy prosody were each 75% consistent with theory and sad prosody was not (25%). Second, in relation to non-emotional prosody, angry, happy, and sad prosodies were consistent with the empirical evidence base, 70%, 90%, and 50%, respectively. A subjective study was conducted wherein 30 adults rated the speech samples. Overall, adults discriminated the intended emotion of the speaker with 92% accuracy. Multiple regression analyses indicated that less than 25% of the significant acoustic patterns for each prosody accounted for variance in perceived emotional intensity. [Work supported by an award to Pamela Cole (NIMH 104547).]

11:20

3aSC10. Vocal emotion identification by older listeners with hearing loss. Huiwen Goy and Frank A. Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, huiwen.goy@psych.ryerson.ca)

A speaker's emotional state is one important type of information carried by the speech signal. Numerous studies have been conducted on younger and older normal-hearing listeners' ability to identify vocal emotions. Much less is known about how hearing loss affects emotion identification, and whether listeners with hearing loss use similar acoustic cues as normal-hearing listeners to identify emotions. The purpose of this study was to evaluate vocal emotion identification performance by younger normal-hearing listeners and older listeners who use hearing aids, and to examine possible acoustic cues used by listeners to distinguish between emotions. The results showed that older listeners with hearing loss performed much worse in a quiet listening environment (38.9%) than young normal-hearing listeners in quiet (84.7%) or in noise (68.4%), and that the use of hearing aids did not improve their performance significantly. Consideration of the pattern of identification errors and the acoustics of the stimuli suggested that the two listener groups relied on different F0-related cues to distinguish emotions: young listeners related on F0 mean and F0 contour, while older listeners relied mainly on F0 mean. Understanding how listeners with hearing loss identify emotion would provide guidance for developing rehabilitative strategies for hearing loss.

11:40

3aSC11. Processing of emotional sounds conveyed by the human voice and musical instruments. Sophie Nolden (Goethe-Univ. Frankfurt am Main, Theodor-W.-Adorno-Platz 6, Frankfurt am Main 60629, Germany, nolden@psych.uni-frankfurt.de), Simon Rigoulot (Int., Lab. for Brain, Music and Sound Res., Montreal, QC, Canada), Pierre Jolicoeur (Univ. of Montreal, Montreal, QC, Canada), and Jorge L. Armony (McGill Univ., Montreal, QC, Canada)

We investigated how emotional sounds are processed by musicians and non-musicians. Expertise through musical training can shape the way emotional music is processed. However, emotional sounds can arise from other sound sources, too, such as vocal expressions like laughter or cries. Musical expertise can also influence the way these emotional sounds are processed, thus, the impact of musical expertise can generalize over a wide range of emotional sounds. A recent study investigated oscillatory brain activity related to the processing of emotional sounds in musicians and non-musicians (Nolden, Rigoulot, Jolicoeur, & Armony, 2017, see also Rigoulot, Pell, & Armony, 2015). Musicians showed greater overall oscillatory brain activity than non-musicians. This was true for emotional music as well as for emotional sounds produced by the human voice, suggesting that expertise in processing one domain generalizes to different domains.

WEDNESDAY MORNING, 7 NOVEMBER 2018 — SALON B (VCC), 8:30 A.M. TO 11:20 A.M.

Session 3aUWa

Underwater Acoustics, Acoustical Oceanography, Animal Bioacoustics, and Physical Acoustics: Biological Effects on Seabed Geoacoustic Properties

Kevin M. Lee, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Megan S. Ballard, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78758

Kelly M. Dorgan, Cochair
Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528

Chair's Introduction—8:30

Invited Papers

8:35

3aUWa1. The trait-based approach as a 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.

8:55

3aUWa2. Effects of burrow and tube construction by infauna on sound propagation through marine sediments. Kelly M. Dorgan, Will M. Ballentine, Grant Lockridge (Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, kdorgan@disl.org), Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX)

Infauna, animals living in marine sediments, modify sediment structure by burrowing, constructing burrows and tubes, and irrigating burrows. These activities can change the bulk porosity and density as well as create heterogeneity in sediment structure. We test the hypothesis that these activities alter sound speed and attenuation in sediments by manipulating homogenized sediments to mimic animal

activities. Specifically, we examine the effects of burrow excavation, burrow wall compaction, burrow irrigation, and construction of tubes from shell hash on sound speed and attenuation at 100, 200, and 400 kHz. Wavelengths corresponding to these frequencies span the size of the burrows constructed, and measurements were conducted at several depths within the upper 10 cm of sediment in which infauna are commonly found. Each of these activities or functions is performed by multiple species of animals that comprise a functional group. Our results will help identify functional groups that have important impacts on sediment acoustics and will be used to interpret field data in which deviations from predicted sound speed and attenuation are correlated with different and diverse communities of infauna.

Contributed Paper

9:15

3aUWa3. Effects of benthic biology on geoacoustic properties of marine sediments. 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), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Infauna dwell in the benthic zone and have the capacity to modify the physical and acoustic properties of the seabed through bioturbation. To investigate such effects, *in situ* measurements of compressional and shear wave speed and attenuation were conducted in Petit Bois Pass, near the mouth of Mobile Bay, Alabama, USA. The measurement system was deployed at multiple locations within the pass, and acoustic measurements were conducted at depths up to 20 cm into the sediment to scan the portion of the seabed where most infauna live. Additionally, diver cores were collected and analyzed for infauna abundance and geotechnical properties, such as porosity and grain size distribution, for comparison with the acoustic data. While sediment geoacoustic models do not explicitly account for effects of biology, they do allow for parameterization of various physical properties like porosity, grain size, or pore fluid viscosity, all of which can be modified by the presence of biological organisms and bioturbation. Results from the *in situ* acoustic measurements and core analysis will be compared with such models to determine if any additional insight into the acoustic effects of infauna can be gained. [Work supported by ONR and ARL:UT IR&D.]

Invited Papers

9:30

3aUWa4. Effects of benthic biology on normal-incidence reflection coefficient measurements. Marcia J. Isakson and Lucas Shearer (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Normal-incidence reflection coefficient measurements have been shown to be sensitive to sediment type based on the structure and amplitude of the returned signal. In contrast to other sediment classification methods such as coring, these measurements can be done quickly and easily over a large area. This type of sediment classification scheme can also be integrated into surveys by single- and multi-beam echo sounders. However, benthic biology can substantially change the normal-incident acoustic return possibly confounding the sediment classification methods while simultaneously providing a method of assessing benthic eco-systems. In this study, a collection of normal-incidence measurements taken in a variety of underwater environments will be analyzed for the effects of benthic biology. The areas surveyed include the Chesapeake Bay, the Arctic, the Western Mediterranean, and Gulf Coast. Each of the datasets will be considered for the presence of benthic biology and its affect on geo-acoustic properties. [Work supported by ONR, Ocean Acoustics.]

9:50–10:05 Break

Contributed Paper

10:05

3aUWa5. Observations of the effects of benthic biology in measurements from the Acoustic Coring System collected in the New England Mud Patch. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Jason D. Chaytor (U.S. Geological Survey, Woods Hole Coastal and Marine Sci. Ctr., Woods Hole, MA), Allen H. Reed (Naval Res. Lab., Stennis Space Ctr., MS), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This paper presents *in situ* measurements of sediment sound speed and attenuation collected in the New England Mud Patch using the Acoustic Coring System (ACS) with comparisons to sound speed measurements from a multi-sensor core logger (MSCL). The ACS uses rod-mounted piezoelectric cylinders mounted below the penetrating tip of a gravity corer to obtain a continuous record of sound speed as a function of depth as the corer penetrates the seabed. The MSCL measurements of the recovered sediment cores were acquired in 2 cm increments. In both the ASC and MSCL measurements, an elevated sound speed is observed in the upper 25 cm of the seabed which is the portion of the seabed where most infauna live. The sediments collected in the cores were analyzed for density, porosity, bulk organic matter, carbonate content, mineral composition, and grain size. Additional surficial sediment samples recovered using a box core and a multi-core were sieved for infauna collection. The organisms were preserved, identified, and classified according to acoustically relevant traits. The acoustic measurements are interpreted in terms of the physical measurements of cores and the abundance of different types of organisms present in the seabed. [Work supported by ONR.]

10:20

3aUWa6. Geoacoustic properties of seagrass-bearing sediments. Gabriel R. Venegas (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Abdullah F. Rahman (Coastal Studies Lab., The Univ. of Texas Rio Grande Valley, Brownsville, TX)

Seagrasses are commonly referred to as "ecosystem engineers" due to their ability to modify their environments to create unique ecosystems believed to be the third-most valuable in the world. While engineering nursery habitats for small invertebrates, fish, and microalgae such as diatoms, they also aid in water filtration and are capable of storing 11% of the ocean's organic carbon each year. Seagrasses mainly grow in shallow salty waters within sandy sediments, but over time can drastically alter the sediment by entrapping silt particles, organic matter and decaying organisms that collect on the seafloor to form an organic-rich mud that incorporates itself with the underlying sand. Anaerobic microbes break down the buried organic matter to produce biogas and more stable organic compounds, which can drastically affect sediment geoacoustic properties. Sediment cores were collected within a *Thalassia testudinum* meadow in the Lower Laguna Madre, TX, and analyzed in 2-cm depth increments for grain size, density, porosity, sound speed, and attenuation from 100 kHz to 300 kHz, and organic carbon content. Results are compared with those from a seagrass-free sediment core to investigate how these valuable "ecosystem engineers" can alter geoacoustical properties of the seabed. [Work supported by ONR and ARL:UT IR&D.]

10:40

3aUWa7. High-frequency acoustic scattering from aquatic plants and seagrass leaf blades. Jay R. Johnson (Mech. Eng., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jean-Pierre Hermand (LISA Environ. HydroAcoust. Lab, Université libre de Bruxelles (ULB), Brussels, Belgium), and Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Aquatic plants and seagrasses play a vital role in the littoral zone and shallow water ecosystems. They are important for carbon sequestration, habitat health, and seabed stabilization. Both aquatic plants and seagrasses are vascular plants with gas-filled internal channels, known as aerenchyma, with complex acoustic responses. Many previous field and laboratory experiments have shown that the acoustic response of vascular aquatic vegetation depends on tissue gas content, leaf blade structure, plant density, and photosynthetic activity, yet forward predictive models for acoustic propagation in aquatic plants and seagrass have not been realized. To help progress towards such a model, laboratory bistatic acoustic scattering measurements at 1 MHz were performed on aquarium raised *Vallisneria spiralis*. The measured results are compared to forward scattering measurements, at the same frequency, performed on naturally collected Mediterranean seagrasses *Posidonia oceanica* and *Cymodocea nodosa*. Microscopic images of leaf blade cross-sections were used to investigate the effect of tissue structure on acoustic response and nascent analytic and numerical models are presented to help interpret the measured results. [Work supported by ARL:UT IR&D, ONR, ONR Global, FNRS.]

11:00

3aUWa8. Effect of the microphytobenthos photosynthesis on seabed backscattering properties. Natalia Gorska, Adam Latala, and Filip Pniewski (Inst. of Oceanogr., Faculty of Oceanogr. and Geography, Univ. of Gdansk, al. Marszalka Pilsudskiego 46, Gdynia 81 - 378, Poland, oceng@ug.edu.pl)

The development of hydroacoustic techniques for benthic habitat classification requires understanding of biological effects on seabed geoacoustic properties. The study addresses only one biological process that could potentially change the properties—photosynthesis of benthic microalgae. The previous investigations demonstrated that the impact could be important in warm southern marine waters of relatively high salinity. This motivated us to study the effect of the microphytobenthos photosynthesis on the backscattering properties of the sandy sediments of the southern Baltic Sea, an area of lower temperature and salinity. The five multiday laboratory experiments, different in hydrophysical or biological conditions, were conducted. The backscatter data were acquired in the small tank with sandy bottom under controlled constant temperature and salinity with simulated "day" and "night" conditions (light/dark (L/D) photocycles). Oxygen content in the water column as well as biological and biooptical parameters were additionally monitored. The diel variations of the backscattered signal energy were analyzed. The study demonstrated the impact of microphytobenthos photosynthesis on the backscattering properties of the marine sediment and its sensitivity to the abiotic (illumination level) and biotic (benthic microalgal biomass and macrozoobenthos bioturbation) factors.

Session 3aUWb

Underwater Acoustics: Topics and Modelling in Underwater Acoustics

Darrell Jackson, Cochair
Applied Physics, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Dale D. Ellis, Cochair
Physics, Mount Allison University, 18 Hugh Allen Drive, Dartmouth, NS B2W 2K8, Canada

Contributed Papers

9:00

3aUWb1. The mutual interface scattering cross section. Darrell Jackson and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

Scattering by the ocean surface or seafloor is often characterized by the interface scattering strength, the decibel equivalent of the scattering cross section per unit solid angle per unit area (referred to subsequently as the interface scattering cross section for brevity). The scattering cross section is a statistical second moment that allows modeling of the mean-square scattered pressure. Use of the interface scattering cross section requires that the incident field can be approximated by a single plane wave, a condition that is not satisfied when the interface is ensonified in the near field of a large array, as in synthetic-aperture sonar (SAS), or when multipath propagation is important as in modeling reverberation. The mutual cross section introduced in this presentation is a generalization of the interface scattering cross section to include ensonification by multiple plane waves. Examples of application to SAS and the ocean waveguide will be shown. In the latter case, it is seen that the mutual cross section can model peaks in the reverberation time series that occur as a result of multipath convergence on the interface. (Work supported by ONR.)

9:15

3aUWb2. Autoproducts and out-of-band acoustic fields in refracting multipath environments with caustics. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., 2010 Automotive Lab, Ann Arbor, MI 48104, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Autoproducts are nonlinear mathematical constructions developed from a given acoustic field with non-zero bandwidth that can—with some limitations—mimic lower and higher frequency acoustic fields lying outside the bandwidth of the given acoustic field. The autoproducts' mimicry of out-of-band fields is often sufficient for successful beamforming and/or source localization at below- or above-band frequencies chosen to correct array sparseness, suppress random scattering, mitigate environmental mismatch, and/or enhance resolution. While the limitations of autoproducts developed from acoustic fields well-described by ray acoustics are understood, what is less studied are the effects of diffraction and refraction. In this presentation, acoustic waveguide environments are defined which contain strong refraction, and form caustics—regions where nearby ray paths cross—which results in a phase shift in the propagating waves. These caustic phase shifts are not found in autoproduct fields and are detrimental for autoproduct-based remote sensing, especially in regions that are reached by multiple rays that have passed through different numbers of caustics. Multiple refracting environments are considered, including n^2-linear and n^2-quadratic sound speed profiles, and the ability of autoproducts to mimic out-of-band fields in these environments is assessed. Implications for autoproduct-based source localization are discussed. [Sponsored by ONR and NSF.]

9:30

3aUWb3. An improved Gaussian beam caustic correction for Bellhop at low frequencies. Diana F. McCammon (McCammon Acoust. Consulting, Waterville, NS, Canada) and Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com)

Gaussian beams are commonly used in ray-tracing to mitigate the effects of caustics and shadow zones. The problem is how to determine the width of these "fuzzy" beams. Porter and Bucker [J. Acoust. Soc. Am. 82, 1349–1359, 1987] proposed a method that expressed the beamwidth and curvature in terms of p and q of the dynamic ray equations. We call q the beamwidth factor. In the Gaussian beam implementation caustics are not caused by the crossing of two rays; rather they occur when the beamwidth factor, which appears in the denominator of the amplitude, becomes small. In practice, this is generally not a problem at high frequencies, but as the frequency gets lower the problem gets more severe. The widely used Bellhop model has a procedure, which "caps" the beamwidth when q becomes too small, but the procedure eventually breaks down at low frequencies. Here, we propose a different cap based on a cylindrical wave front converging to the focal point of a caustic. The various caps are compared with the "exact" normal mode solution for a shallow-water upward refracting environment, illustrating how the new cap provides better reduction of the caustic anomalies.

9:45

3aUWb4. Modeling of the power delay profile for underwater acoustic channels—Effects of out-of-plane scattering and reverberation. Trond Jenserud (Norwegian Defence Res. Establishment (FFI), Karljohansvern, Horten NO-3191, Norway, trond.jenserud@ffi.no) and Sven Ivansson (Swedish Defence Res. Agency (FOI), Kista, Sweden)

Long reverberation tails seem to be a common feature of shallow water acoustic channel impulse responses. We show examples from two significantly different environments along the Norwegian coast, and demonstrate that 2-D propagation modeling fails to reproduce the long tails observed. The continuous nature of the impulse responses and the low correlation and slow decay rates of the tails point towards reverberation as an important mechanism behind the observations. The effects of reverberation and out-of-plane scattering are then included in the modeling by employing the 3-D ray-based model Rev3D. Scattering from the sea surface and sea bottom are described by scattering-strength functions including azimuthal variation. The modeling results indicate that the long reverberation tails can be explained by 3-D out-of-plane scattering. Although the measured levels are not perfectly matched, the importance of including reverberation in the modeling is clearly demonstrated.

10:00–10:15 Break

10:15

3aUWb5. Model and data comparison for a three-dimensional, finite-difference, time-domain solver in Sequim Bay. Erin C. Hafla (Civil Eng., Montana State Univ., 205 Cobleigh Hall, Bozeman, MT 59717-3900, erinhafla@gmail.com), Erick Johnson (Mech. Eng., Montana State Univ., Bozeman, MT), Jesse Roberts (Sandia National Lab., Albuquerque, NM), and Kaustubha Raghukumar (Integral Consulting, Inc., Santa Cruz, CA)

Paracousti is a three-dimensional finite-difference, time-domain solution to the governing velocity-pressure equations. This program is directed at modeling sound propagation generated by marine hydrokinetic (MHK) sources in an ocean environment. It is capable of modeling complex, multi-frequency sources propagating through water and soil that have spatially varying sound speeds, bathymetry, and bed composition. Experimental sound data collected at Sequim Bay in Washington, USA, during the winter of 2017 is compared against several simulations modeled within Paracousti for a range of frequencies and receiver locations. This measurement campaign recorded ambient noise data and the sound from a source producing three-second long, sinusoidal pulses between 20 and 5,000 Hz at a depth of 3 m. Additionally, bathymetric, sanity, and temperature data for the bay were collected in order to calculate the sound speed. Data were recorded at six locations ranging in distance between 10 and 1,000 m from the source by stationary buoys. Each simulation was created to model the collected source profiles and has a total depth of 80 m, with the average soil depth occurring at 23 m, and compared via transmission losses.

10:30

3aUWb6. Investigation of property losses in single crystal transduction devices. Michael Warnock and Thomas R. Howarth (Naval Undersea Warfare Ctr., 25 Merton Rd., Newport, RI 02840, michaelwarnock4@gmail.com)

With the advent of relaxor single crystal materials, such as lead indium niobate-lead magnesium niobate-lead titanate (PIN-PMN-PT) and lead magnesium niobate-lead titanate (PMN-PT), it was discovered that the highly oriented crystal structure of the materials leads to large values of dielectric and piezoelectric properties favorable to acoustic applications. When utilized in transduction devices, single crystal materials have demonstrated better acoustic performances compared to traditional designs. However, during construction of single crystal transducers, a capacitance loss is measured after mechanically bonding the crystal to a stiff backing material. The lower capacitance suggests a decrease in the effective permittivity, decreasing favorable piezoelectric constants for acoustic performance. In this study, the loss in effective permittivity was modeled using a commercial finite element analysis software. We present simulations used to demonstrate the loss in effective permittivity and compare with experimental measurements. Additionally, finite element simulations of several design considerations that can be utilized by transducer designers to minimize the loss in the effective permittivity are presented.

10:45

3aUWb7. Soundscape characteristics in Southern Resident Killer Whale critical habitats. Svein Vagle (Fisheries and Oceans Canada, 9850 West Saanich Rd., Sidney, BC V8L4B2, Canada, Svein.Vagle@dfo-mpo.gc.ca), Caitlin O'Neill (Fisheries and Oceans Canada, Victoria, BC, Canada), Sheila Thornton, and Harald Yurk (Fisheries and Ocean Canada, West Vancouver, BC, Canada)

The Southern Resident Killer whales (Orcinus orca) (SRKW) are an endangered group of orcas with current range of Pacific North East from California to Northern British Columbia and spend most of the summer months in and around the Salish Sea. This group of mammals feed primarily on fish, are very local, and live in tight-knit family units called pods. July 1 2017 census reported 77 animals; which now has been reduced to 76 by a more recent death. Anthropogenic underwater noise, primarily from commercial and recreational vessels is suspected to have detrimental effects on these whales. Here, we present results from a seven month, continuous sound recording (125 kHz acoustic bandwidth), whale centered study to monitor and interpret different soundscapes in SRKW critical habitats. The six different locations studied have significantly different acoustic transmission characteristics and cover open ocean areas, where natural sound spectral levels are high, to areas where anthropogenic noise-sources dominate. Possible implications of these different characteristics on the ability of the orcas to communicate and find prey are discussed. [Work funded by the Ocean Protection Plan (OPP) of the Government of Canada.]

11:00

3aUWb8. Grid-free compressive DOA estimation via alternating projection. 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), Mark Wagner, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

On a discrete angular grid, the compressive sensing (CS) reconstruction degrades due to basis mismatch when the direction-of-arrivals(DOAs) do not coincide with the angular directions on the grid. To overcome this, we solve the DOA estimation in the grid-free angular domain. A method for solving the grid-free CS based beamforming problem is presented. The grid-free CS beamforming involves using a convex relaxation of a sparsity exploiting method and solving its semidefinite characterization. Current semi-definite programming solvers have high computational complexity for problems formed from typical large datasets. We solve the non-convex formulation of the grid-free CS beamforming optimization problem with alternating projections (AP). The AP based grid-free CS beamforming can solve large scale problems with hundreds of observations in a few seconds. We compare and contrast our algorithm with CS based beamforming as well as conventional beamforming.

11:15

3aUWb9. Validation of airgun array modelled source signatures. Craig McPherson (JASCO Appl. Sci., Unit 4, 61-63 Steel St., Capalaba, QLD 4157, Australia, craig.mcpherson@jasco.com), Alexander O. MacGillivray (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), and Edward Hagar (Polarcus Asia Pacific, Singapore, Singapore)

Characterizing the source levels from seismic airgun arrays to increase confidence in model accuracy has been identified as a requirement by the scientific and regulatory community. To comply with regulatory requirements in Australia, a measurement program was conducted to validate the source signature predictions of JASCO's Airgun Array Source Model. The validation program measured a range of commercial airgun arrays, including a $2380in^3$ array, and was conducted in 80m of water with array passing directly over the recorder on the seafloor. For the $2380in^3$ array, the maximum measured PK was 220.62 dBre1μPa. Due to the proximity of the measurements, interesting features such as ghost cavitation and airgun recharge were observed in the data. The measurement study results were used to validate the modelled far-field source levels through a comparison between the measured received sound-levels and predicted received sound-levels at a real receiver point in the far-field of the source. The predictions were made using a wavenumber integral model coupled to the airgun source model. The program measured received sound levels in the endfire, broadside and vertical directions, and as the results showed good agreement with the modelling results, provided a validation of the complete modelled source signatures for the array.

Session 3pAA

Architectural Acoustics: Absorption, Diffusion, and Insulation

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Ian B. Hoffman, Cochair
Peabody Institute: Johns Hopkins Univ., 1 E Mount Vernon Place, Baltimore, MD 21202

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Chair's Introduction—1:00

Contributed Papers

1:05

3pAA1. Cross-stitched fabric as potential sound absorber for room acoustics applications. Xiaoning Tang and Xiong Yan (College of Textiles, Donghua Univ., Renmin North Rd. 2999, Shanghai 201620, China, tangxn123@163.com)

Textiles materials are widely used in noise and vibration control engineering. A thin fabric absorber consists of an air cavity and a backing rigid wall, and its main applications include ceiling, curtain, and sound barrier, etc. The main aim of this work is to study the acoustical and sound absorption properties of cross-stitched fabric, which has a similar structure with micro-perforated panel. The sound absorption coefficients of cross-stitched fabrics with different specifications were measured by impedance tube method, where the air cavity distance is 1 cm, 2 cm, and 3 cm, respectively. These distances are suitable to the binding and layout design of cross-stitched fabric painting in practical applications. As industrial textiles, cross-stitched fabric can be made into diversity of artistic patterns through manual or mechanized embroidery technology. This work has also studied the effects of multifarious embroidered patterns on the sound absorption coefficients of fabrics with different air cavity distance. In summary, cross-stitched fabric with exquisite patterns is expected to be a promising sound absorber for room acoustics.

1:20

3pAA2. Wave-based modeling of sound insulation of a double layer finite panel with different infilling materials between two reverberant rooms. Fangliang Chen, Yihe Huang, and Tejav DeGanyar (Virtual Construction Lab, Schuco-USA, 260 West 39th St., Ste. 1801, New York, NY 10018, fchen@schuco-usa.com)

A wave-based model is presented in this study to investigate the direct sound transmission through a double layer panel infilled with different materials and embedded between two rooms. The modal behavior of both the rooms and elastic finite layers are considered; accordingly, the full vibroacoustic coupling between rooms, panels, and air cavity are accounted for. To validate the presented model, experiments on a double layer finite panel with two aluminum face sheets infilled with different absorption materials between two reverberant rooms are conducted at a certificated testing chamber. Comparisons between the results predicted by the wave-based model and those obtained by INSUL are presented as well. Comprehensive parametric studies are further conducted to study the effects of different design parameters on the sound transmission loss of a double layer panel., such as different filling materials, panel locations in the partition wall, cavity dimensions, boundary conditions of the panel embedded in the wall, damping factor of the elastic layers, panel dimension, and aspect ratio. Findings from the parametric studies will provide an important practice value to guide the design and assembling of a desired double layer panel to achieve an optimized performance of sound insulation between two adjacent rooms.

1:35

3pAA3. Sound absorption through PDMS-based flexible perforated plate backed by a dual layer cavity. Sanjay Kumar, Pulak Bhushan, and Shantanu Bhattacharya (Dept. of Mech. Eng., Indian Inst. of Technol. Kanpur, Microsystems Fabrication Lab., Kanpur, Uttar Pradesh 208016, India, sanjay21505@gmail.com)

A structure consisting of a perforated plate backed by an air cavity is a widely used structure for sound absorption in the low-to-high frequency range. In this study, we propose a sound absorbing metastructure composed of a polydimethylsiloxane (PDMS)-based flexible perforated plate backed by a layer of multiple hexagonal unit cells. Each unit cell consists of a dual cavity connected by a rigid perforated plate. The proposed structure allows for enhanced vibroacoustic coupling between the perforated plate and the back cavity. First, the structure is theoretically simulated using the finite element based analysis. The results elicit that the shape and size of the back cavity significantly alter the sound absorption coefficient and the resonant frequency of the structure. The geometrical parameters of the metastructure were further optimized using an artificial intelligence-based technique. Using the optimized geometrical parameters, the back cavity was printed using selective laser sintering (SLS) and a PDMS perforated plate has adhered on top of it. The experimental results demonstrate a potential sound absorption metastructure which can be utilized for multiple applications.

1:50

3pAA4. A diffuser language: Designing quadratic residue diffuser arrays with shape grammars. Jonathan Dessi-Olive and Timothy Hsu (Architecture, Georgia Inst. of Technol., School of Architecture - Georgia Tech, 245 4th St., NW, Ste. 351, Atlanta, GA 30332, jdo@design.gatech.edu)

This paper presents research on a rule-based approach to designing creative acoustic diffuser arrays. A shape grammar-influenced design method is specified that uses shape rules to recursively design arrays of quadratic residue diffusers (QRD) in a way that is neither mechanical nor deterministic. Shape grammar-based generative systems have already been shown to be capable of being used in architecture and engineering to create languages of

functional objects but have not yet been extended into the realm of acoustics. The grammar presented by this paper produces a QRD-based acoustic diffusion language that breaks habits in QRD deploying techniques, which is lacking outside of using known equations that give known forms. The grammar can include different design frequencies and open the possibilities of different and non-uniform, intentional diffusion treatment. This paper will specify and demonstrate the formation rules, and discuss on how in combination with material specifications, this design method can generate broadband QRD arrays that address both performative and aesthetic criteria in new and surprising ways.

2:05

3pAA5. Acoustic impedance control for reproducing sound environment by constructing virtual wall. Eiji Yokota and Seiichiro Katsura (Integrated Design Eng., Keio Univ., 3-14-1, Hiyoshi, Kohoku, Yokohama, Japan, Yokohama, Kanagawa 223-8522, Japan, yokota@katsura.sd.keio.ac.jp)

In order to obtain high presence, it is necessary to reproduce the sound environment of remote places. Conventionally, sound environments at remote places are reproduced by recreating a sound signal at that place. In this study, an acoustic technique that reproduces sound environments of remote places by recreating the dynamics of the environment at that place is proposed. In the proposed acoustic technique, the desired sound environment is attained by making listeners on the spot feel as if there were a wall that reflects and absorbs the sound waves. By controlling the acoustic impedance at the position where the wall is constructed, the wall is virtually built. In addition, the acoustic impedance including the virtual viscosity and elasticity that can be arbitrarily determined at the position where the wall constructed is set. As a result, by deciding the acoustic output from the secondary sound source so as to attain the desired acoustic impedance, the wall is virtually constructed to reproduce the desired sound environment.

2:20

3pAA6. Comparative study of the sound absorption coefficient of two absorbent materials with two empirical methods for low frequency. Nelson L. Legat, Claudia C. Silva, Nilson Barbieri, and Key F. Lima (PUCPR, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com)

The equations of Delany and Bazley have been widely used in the study of the behavior of sound waves inside porous materials since 1970. However, as is known, the validity of these equations for low frequencies can produce physically inconsistent results. This work intends to evaluate the sound absorption coefficient of two types of absorbent material, one porous and one fibrous, with the Allard and Champoux's equations presented in 1992. These equations have the purpose of improving acoustic properties characterization at a low frequency. The precision of the results obtained through the two empirical methods is verified with an experimental evaluation using the modified two cavity method. The measurements are made on the impedance tube with the samples arranged in two different positions displaced from the rigid termination. Porous samples evaluated by this work were common foam and the fibrous were coconut coir fibers.

2:35

3pAA7. The acoustics of the "Witches Valley." Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Gino Iannace, and Amelia Trematerra (Seconda Universita' di Napoli, Aversa, Caserta, Italy)

According to legend, witches were women with magical knowledge that with the use of herbs could cure illnesses or perform spells. In 1639, the book "De Nuce Maga Beneventana" was published and described how the place where the "witches" gathered was an area south in South Italy in a long, narrow gorge with high rock walls, called the "Stretto di Barba." The legend of the witches originated with the invasion of southern Italy by the Longbards who, according to legend, under a walnut tree performed sacred rituals with dances and chants accompanied by the rhythm of drums. To make the rituals more effective, mysterious, and emotionally involving for the people take, it was necessary to amplify the sounds generated. This could only take place in the presence of a narrow gorge with flat and parallel rock walls where the reflecting of the sound generated echoes. The aim of this paper is to verify whether the "Stretto di Barba" has the acoustic characteristics that create echoes and sound amplification effects to confirm the legend that the Longbards performed their rituals here.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 SHAUGHNESSY (FE), 1:00 P.M. TO 3:30 P.M.

Session 3pAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species II

Thomas F. Norris, Cochair
Bio-Waves Inc., 517 Cornish Dr., Encinitas, CA 92024

Tiago A. Marques, Cochair
School of Mathematics and Statistics, University of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews KY16 9 LZ, United Kingdom

Chair's Introduction—1:00

Invited Papers

1:05

3pAB1. Sound localization for estimating population densities of birds: Progress, challenges, and opportunities. Richard W. Hedley and Erin M. Bayne (Biological Sci., Univ. of Alberta, 11335 SK Dr. NW, Edmonton, AB T6G 2M9, Canada, rhedley@ualberta.ca)

Autonomous recording units (ARUs) are seeing rapid adoption as a research tool to estimate population densities of birds. One challenge for estimating population density is that the distance to the vocalizing bird is unknown. An emphasis has been placed on assessing the detection radius for different species on ARUs, and more recent work has had success relating sound amplitude to distance. We argue, however, that the true variable of interest is not density, per se, but the relationship between density and habitat type or other variables. In an ideal world, ARUs would collect information on the absolute location of each sound source, rather than the number of each species within earshot of the microphone, since the detection radius of an ARU commonly encompasses multiple habitat types. Sound localization is a promising solution to this problem, with some localization methods allowing sound sources to be pinpointed within 1m of their true location. We will present localization results comprising several thousand songs from several species in the Boreal forest, to highlight how localization data can be translated into estimates of population density. We will also highlight several challenges and opportunities for future contributions in the field of sound localization.

1:30

3pAB2. Detection probability of Cuvier's beaked whale clicks from a glider and a deep-water float. Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@oregonstate.edu), Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR), Jay Barlow (Marine Mammal and Turtle Div., Southwest Fisheries Sci. Ctr., NOAA, La Jolla, CA), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St Andrews, United Kingdom), and Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithica, NY)

Work is being conducted to estimate marine mammal density and abundance from slow-moving, passive acoustically equipped underwater gliders and deep-water floats. We deployed five drifting acoustic spar buoy recorders (DASBRs) simultaneously with a seaglider and QUEphone float in the Catalina Basin of Southern California in 2016 to estimate the probability of detecting Cuvier's beaked whale echolocation clicks from the glider and float. The DASBRs successfully localized and tracked individual whales, though a limited number of encounters prohibited estimation of detection probability through a trial-based method. We explored using a spatially explicit capture recapture (SECR) approach. The SECR analysis was modified to account for the non-static array and we explored sample size requirements for SECR using simulations. During tracked dives, over 200 one-minute time bins contained echolocation clicks on at least one DASBR. Of these, 10 and 34 bins contained clicks recorded on the glider and float, respectively, and were used to make preliminary estimates of detection probability as a function of range. Detection probability estimation is a key step towards animal density estimation; other information such as encounter rate and click production rates are required to estimate animal density, which will form the next part of this work.

1:55

3pAB3. Estimating densities of terrestrial wildlife using passive acoustic recordings: A pragmatic approach using paired human observations. Steven Van Wilgenburg (Environment and Climate Change Canada, Canadian Wildlife Service, 115 Perimeter Rd., Prairie and Northern Wildlife Res. Ctr., Saskatoon, SK S7N0X4, Canada, steven.vanwilgenburg@Canada.ca), Péter Sólymos (AB Biodiversity Monitoring Inst., Edmonton, AB, Canada), Kevin Kardynal (Environment and Climate Change Canada, Sci. & Technol. Branch, Saskatoon, SK, Canada), and Matthew D. Frey (Univ. of SK, Saskatoon, SK, Canada)

The use of passive using acoustic monitoring is a rapidly growing field in terrestrial ecology, particularly in ornithology. Much recent work has focused on using recording technologies for large-scale occupancy monitoring, while local scale studies have focused on acoustic localization for density estimation. Density estimation offers numerous advantages for answering key conservation questions but may seem impractical to employ at national or continental scales using current acoustic localization techniques. Here, we describe how paired sampling can be used in conjunction with generalized linear (GLM) or generalized linear mixed models (GLMM) to estimate correction factors (δ) to derive density estimates from single acoustic recorders, provided that paired human observers conduct distance estimation. Thus, our approach provides an alternative to more complicated and expensive methods. We discuss the advantages and disadvantages of our approach and highlight the context in which we see various existing or emerging acoustic density estimation methods being used in the terrestrial environment, given the nature and scale of the ecological question(s) being asked.

2:20

3pAB4. An evaluation of density estimation methods using a multiyear dataset of localized bowhead whale calls. Katherine H. Kim (Greeneridge Sci., Inc., 90 Arnold Pl, Ste D, Santa Barbara, CA 93117, khkim@greeneridge.com), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Tiago A. Marques (Centro de Estatística e Aplicações, Faculdade de Ciências da Universidade de Lisboa, St. Andrews, Fife, United Kingdom), Danielle Harris, Cornelia S. Oedekoven (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Gisela Cheoo (Departamento de Estatística e Investigação Operacional, Faculdade de Ciências da Universidade de Lisboa, Lisboa, Portugal), Aaron Thode (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), and Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, CA)

For eight years (2007–2014), Greeneridge Sciences deployed, collected, and analyzed an immense passive acoustic dataset to study the response of bowhead whales (*Balaena mysticetus*) to oil exploration activities along a 280 km swath of the Alaskan Arctic continental shelf. Each year, up to 40 Directional Autonomous Seafloor Acoustic Recorders (DASARs) detected bowhead calls during the whales' annual fall migration, resulting in over 2.4 million localized calls over the study period. These enormous sample sizes, combined with the unique localization capability of the DASARs, have enabled the investigation and cross-validation of several acoustic density estimation (DE) methods. Here, we compared results among three density estimation methods: (i) direct census, (ii) distance sampling, and (iii) spatially explicit capture recapture (SECR). In the course of our investigation, we encountered and addressed two challenges: (i) how to apply point transect theory to distributed array systems, and (ii) how to implement SECR in situations where detections between sensors are not statistically independent or where false positives exist on single sensors. These density estimates were then used to estimate relative abundance of the Western Arctic bowhead whale population over multiple years. [Work sponsored by the Joint Industry Programme.]

Contributed Papers

2:45

3pAB5. Density estimation using ocean-bottom seismometers in Cascadia. David K. Mellinger (Coop. Inst.for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

The Cascadia Initiative placed a large number of ocean-bottom seismic sensors off the coast of the US Pacific Northwest. These sensors recorded pulsed sounds from fin whales, recorded in conditions that in many cases make it impossible to resolve the sounds from individual whales. The sounds from multiple whales, however, combine to make a distinct rise in spectral energy in the 15–30 Hz band. This received energy can be used to estimate the population density of fin whales in the vicinity of each sensor over the 4-year duration of the project. Difficulties addressed include the variable rate and intensity of fin whale vocalizations, acoustic properties of the seafloor, and differences between seismic and acoustic sensors. [Funding from ONR.]

3:00

3pAB6. Can whistles be used to estimate dolphin abundance? Carmen Bazúa Durán (Física, UNAM, Facultad de Ciencias, Circuito Exterior s/n, Ciudad Universitaria, México, D.F. 04510, Mexico, bazua@unam.mx)

Dolphin whistles have been studied extensively for both wild and captive animals. They are used in the communication between individuals, to maintain contact within individuals of a herd, and to coordinate herd movements. However, little is still known on how whistles can be used to ascribe individuals. Therefore, the present study is focused on determining the stereotypy of the most frequently emitted whistles by captive bottlenose dolphins,*Tursiops truncatus*, housed in two different marine parks. Stereotypy was computed by changing the similarity index while classifying whistles into whistle types using Matlab BELUGA and ArtWARP. Signature whistles are an individual distinction and have a similarity index greater than 95%, therefore, allowing to assess the minimum number of whistling dolphins in a pod that emit signature whistles. This is specially important in the wild, where in some occasions, it is very difficult to assess how many individuals are present. Thus this very simple method will be useful to quantify the number of signature whistles from underwater recordings, and to relate it with the possible number of dolphins present. It is necessary to implement methods like this one to better understand how dolphins are using whistles, since acoustic communication is the most important sense in dolphin species. [Work supported by PAPIIT&PASPA-UNAM.]

3:15

3pAB7. Acoustic versus visual monitoring of Cetaceans in coastal habitats. Benjamin T. Hendricks (Elec. and Comput. Eng., Univ. of Victoria, 3800 Finnerty Rd., Eng. Office Wing, Rm. 448, Victoria, BC V8P 5C2, Canada, hendrick@uvic.ca), Eric M. Keen (Marine Ecology and Telemetry Res., Seabeck, WA), Janie L. Wray (North Coast Cetacean Society , Hartley Bay, BC, Canada), Hussein M. Alidina (Oceans Program, World Wildlife Fund - Canada, Victoria, BC, Canada), Chris R. Picard (Gitga'at Oceans and Lands Dept., Hartley Bay, BC, Canada), Hermann Meuter (none, Hartley Bay, BC, Canada), and T. Aaron Gulliver (Elec. and Comput. Eng., Univ. of Victoria, Victoria, BC, Canada)

Monitoring the aquatic soundscape provides critical information about marine mammal habitat use. It is essential for developing mitigation strategies for areas with expanding marine shipping and industrial activity. A long baseline hydrophone array has been installed in Squally Channel, a culturally, ecologically, and economically important marine environment in northern British Columbia, Canada. The array consists of four synchronized bottom-mounted hydrophones that permanently record and radio-transmit data to a land-based laboratory in real-time. The acoustic monitoring is supplemented with a land-based visual observatory that oversees the same area of approximately 200 km^2. Automated detectors have been developed for vocalizations of humpback whales, orcas, and fin whales. Acoustic data and visual surveys were analyzed for a period of more than 100 days between 2017 and 2018. We present an overview of the acoustic detection functions and their performance by call type and summarize visual survey procedures. Cetacean activity derived from automated acoustic detections and visual observations is analyzed. Finally, acoustic and visual survey methods are compared to assess a) the fraction to which different species are acoustically active while in the area, and b) the effectiveness of both acoustic and visual monitoring efforts for the purpose of cetacean monitoring.

Session 3pBAa

Biomedical Acoustics and Physical Acoustics: Bubble Trouble in Therapeutic Ultrasound II

Christy K. Holland, Cochair
Internal Medicine, Division of Cardiovascular Diseases and Biomedical Engineering Program, University of Cincinnati, Cardiovascular Center Room 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Klazina Kooiman, Cochair
Thoraxcenter, Dept. of Biomedical Engineering, Erasmus MC, P.O. Box 2040, Room Ee2302, Rotterdam 3000 CA, Netherlands

Invited Paper

1:00

3pBAa1. Advances in controlled transcranial bubble-mediated drug delivery and opportunities for transvertebral therapy. Meaghan A. O'Reilly, Ryan M. Jones, Lulu Deng, Kogee Leung, Stecia-Marie P. Fletcher, Rui Xu, and Kullervo Hynynen (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. C713, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

The treatment of central nervous system (CNS) disorders is hindered by the presence of specialized barriers that maintain the privileged CNS environment and in doing so restrict the transport of molecules from the vascular compartment to the parenchyma. The ability of ultrasound in combination with intravenously administered microbubbles to transiently open the Blood-Brain Barrier (BBB) to permit the delivery of therapeutic agents is well established in preclinical models and has reached the stage of clinical investigations. Although the methods are not yet as developed as those for BBB opening, ultrasound can also modify the Blood-Spinal Cord Barrier (BSCB) to facilitate drug delivery. Both treatments have immense potential to revolutionize the treatment of CNS disorders, but their widespread clinical adoption hinges on the establishment of robust methods to deliver, monitor and control the exposures to ensure safe and effective treatments in these sensitive tissues. This talk will present recent advances in methods for brain therapy, including three-dimensional transcranial microbubble mapping for treatment control, and will also present new preclinical findings in BSCB opening and approaches for controlled transvertebral focusing at clinical scale.

Contributed Papers

1:20

3pBAa2. Lipid-shelled microbubbles for ultrasound-triggered release of xenon. Himanshu Shekhar, Arunkumar Palaniappan (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Tao Peng, Melanie R. Moody, Shaoling Huang (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr. at Houston, Houston, TX), Kevin J. Haworth (Depart. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr. at Houston, Houston, TX), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Xenon is a cellular protectant shown to stabilize or reduce neurologic injury in stroke. The goal of this work was to develop lipid-shelled microbubbles for ultrasound-triggered xenon release. Microbubbles loaded with either xenon alone (Xe-MB) or xenon and octafluoropropane (Xe-OFP-MB) were synthesized by high-shear mixing lipids with either 100% xenon, or 90% (v/v) xenon and 10% (v/v) octafluoropropane. The size distribution and the attenuation coefficient of Xe-MB and Xe-OFP-MB were measured using a Coulter counter and a broadband attenuation spectroscopy system, respectively. Gas chromatography/mass spectrometry (GC/MS) was performed to assess the dose and stability of encapsulated xenon. The feasibility of xenon release using 5-MHz pulsed Doppler ultrasound and 220-kHz pulsed ultrasound was tested by ultrasound attenuation spectroscopy. Co-encapsulation of OFP increased the number density, attenuation coefficient, and temporal stability of microbubbles. The GC/MS measurements revealed that 143 ± 20 μL/mg and 131 ± 33 μL/mg of xenon was loaded in Xe-MB and Xe-OFP-MB, respectively. Xe-MB and Xe-OFP-MB retained $54 \pm 11\%$ and $66 \pm 1\%$ of the xenon payload within 15 min of activation, respectively. Attenuation measurements confirmed ultrasound-triggered xenon release. These results suggest that lipid-shelled microbubbles with OFP could serve as ultrasound-triggered xenon delivery agents to attenuate cellular breakdown.

1:35

3pBAa3. Investigating the role of lipid transfer in sonoporation. Miles Aron, Oliver Vince, Michael Gray, and Eleanor P. Stride (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The permeabilisation of cell membranes following exposure to microbubbles and ultrasound has considerable potential for therapeutic delivery. Recent studies have demonstrated that transfer of material takes place between phospholipid-coated microbubbles and cell membranes. The aim of this study was to investigate the impact of this transfer on membrane properties and cell permeability. Microbubbles were prepared with matched concentrations and size distributions from a series of formulations utilising phospholipids with differing molecular characteristics. Quantitative fluorescence microscopy techniques were used to quantify changes in the molecular packing, viscosity, and permeability to model drugs of cancer cells under mild ultrasound exposure (15 s of 1 MHz centre frequency, 125 kPa peak negative pressure, 1 kHz pulse repetition frequency, and 10% duty cycle). The molecular "packing" of cancer cell membranes was found to be

significantly affected by the transfer of both phospholipids and other microbubble coating constituents. This was particularly pronounced in the case of microbubbles containing a specific class of lipids, which promoted a ~4-fold increase in model drug uptake following ultrasound exposure. Similar results were observed in an investigation of permeability in an *in vitro* blood brain barrier model suggesting that microbubble composition may play a key role in the efficacy of ultrasound mediated delivery.

1:50

3pBAa4. Real-time imaging-guided microbubble mediated high intensity focused ultrasound heating in an *ex-vivo* machine-perfused pig liver. Dingjie Suo, Eric Juang, Sara Keller, and Michael Averkiou (Dept. of Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Seattle, WA 98105, dsuo@uw.edu)

High intensity focused ultrasound (HIFU) is used clinically for the ablation of prostate and liver tumors. More recently, HIFU has been used in the brain for locally controlled thermal treatment of nonessential tremors. However, the thermal lesion location is restricted to a small area in the tissue due to geometric and boundary limitations (avoiding skin and bone burns). Bubble-enhanced heating (BEH) allows for the use of lower power and can generate spatially controlled thermal lesions in regions not currently accessible with conventional HIFU. We have studied BEH in vitro and in a machine perfused pig liver (MPL). The MPL presents a versatile platform for perfusion and BEH studies because it is clinically relevant and allows for delivery of microbubbles under ultrasound imaging. We first applied conventional HIFU and BEH in vitro, in an enclosure with acoustic windows contaiing glycerin and microbubbles while monitoring the temperature with a fine wire thermocouple. A focused 1 MHz transducer and various clinically approved microbubbles were used. Then we applied BEH on the MPL by introducing microbubbles under image guidance. With this initial study, we have observed that BEH has resulted in 5-10x energy reduction for the formation of a reference lesion in vitro, and 3-6x in the MPL.

2:05

3pBAa5. High-precision acoustic measurements of the non-linear dilatational elasticity of phospholipid-coated monodisperse microbubbles. Tim Segers (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, Enschede, Netherlands), Emmanuel Gaud (Bracco Suisse S.A., Geneva, Switzerland), Michel Versluis (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl), and Peter Frinking (Bracco Suisse S.A., Geneva, Switzerland)

The acoustic response of phospholipid-coated microbubble ultrasound contrast agents (UCA) is dramatically affected by their stabilizing shell. The interfacial shell elasticity increases the resonance frequency, the shell viscosity increases damping, and its nonlinear behavior increases the generation of harmonic echoes that are routinely used in contrast-enhanced ultrasound imaging. To date, the surface area-dependent interfacial properties of the phospholipid coating have never been measured due to the extremely short time scales of the MHz frequencies at which the microscopic bubbles are driven. Here, we present, high-precision acoustic measurements of the dilatational nonlinear viscoelastic shell properties of phospholipid-coated microbubbles. These highly accurate measurements are now accessible by tuning the surface dilatation of well-controlled monodisperse bubble suspensions through the ambient pressure. Upon compression, the shell elasticity of bubbles coated with DPPC and DPPE-PEG5000 was found to increase up to an elasticity of 0.6 N/m after which the monolayer collapses and the elasticity vanishes. During bubble expansion, the elasticity drops monotonically in two stages, first to an elasticity of 0.35 N/m, and then more rapidly to zero. Integration of the elasticity vs. surface area curves showed, indeed, that a phospholipid-coated microbubble is in a tensionless state upon compression, and that it reaches the interfacial tension of the surrounding medium upon expansion.

2:20

3pBAa6. Antibody targeted ultrasound-responsive nanodroplets for the therapy of brain metastases: A pre-clinical study. Oliver Vince, Luca Bau (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. campus Res. Bldg., Oxford, Oxfordshire OX 3 7dq, United Kingdom, oliver.vince@worc.ox.ac.uk), Sarah Peeters (Oxford Inst. for Radiation Oncology, Univ. of Oxford, Oxford, United Kingdom), Michael Gray, Luke Richards (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Sean Smart, Nicola Sibson (Oxford Inst. for Radiation Oncology, Univ. of Oxford, Oxford, United Kingdom), and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Metastatic tumors in the brain represent one of the leading causes of death from cancer with current treatments being largely ineffective and/or associated with significant side effects due to their lack of targeting. Conjugating MRI contrast agents with a monoclonal antibody for VCAM1 (anti-VCAM1) has previously allowed detection of brain tumor volumes two to three orders of magnitude smaller than those volumes currently detectable clinically. In this study, a novel magnetic and acoustically responsive phospholipid-stabilised nanodroplet formulation has been conjugated with anti-VCAM-1. Preliminary *in vivo* tests have shown that these anti-VCAM-1 nanodroplets can be successfully targeted to both inflamed areas of the brain and metastatic brain tumours. Acoustic droplet vaporisation of the anti-VCAM1 droplets, confirmed by passive cavitation detection, was also shown to cause blood brain barrier permeabilisation *in vivo*. Extensive *in vitro* characterisation of the potential of these nanodroplets to target brain metastases using antibody and magnetic targeting, along with the ultrasound conditions required for various therapeutic effects has been completed and has been found to agree well with mathematical modelling. The implications of these findings and the plans for future investigations into the therapeutic potential of these targeted and ultrasound responsive agents will be discussed.

2:35

3pBAa7. High-speed observations of Acoustic Cluster Therapy "activation" and "therapy". Jae Hee Song (CavLab, Medical and Industrial Ultrason., Univ. of Glasgow, Glasgow, United Kingdom), Andrew Healey (Phoenix Solutions, Oslo, Norway), and Paul Prentice (CavLab, Medical and Industrial Ultrason., Univ. of Glasgow, Ninewells Hospital, Dundee DD1 9SY, United Kingdom, paul.prentice@glasgow.ac.uk)

Acoustic Cluster Therapy (ACT^{TM}) is a novel and versatile drug delivery platform, based on electrostatically bound clusters of negatively charged contrast agent microbubbles (Sonazoid^{TM}, GE Healthcare) and positively charged oil microdroplets, for co-administration with drug to the vasculature. Preliminary *in-vitro* high-speed observations at up to 10 million frames per second of both "activation" (exposure of clusters to 2.19 MHz focused ultrasound, at mechanical index, MI $\approx$ 0.15) and "therapy" (exposure of the resulting vapour bubbles, to 675 kHz with MI ranging from 0.1 to 0.4) will be presented. At low MI, the therapy exposure induces quasi-linear oscillations of the vapour bubble, with higher MIs inducing high-order surface wave activity. In the latter case, the amplitude of the non-spherical oscillations increase during the exposure until involutions neck, generating intra-bubble droplets that traverse the bubble core at speeds of up to several m/s. Possible mechanisms for activation and therapy will be discussed, based on the observations. [This research was supported by ERC Starting Grant TheraCav, 336189, and Glasgow Knowledge Exchange project DefACT.]

Session 3pBAb

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

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

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Contributed Papers

1:00

3pBAb1. Study on the ultrasonic wave convergence in the medium with bone. Masaya Saeki, Leslie V. Bustamante Diaz (Dept. of Elec. and Electronics Eng., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe-shi, Kyoto 610-0321, Japan, ctwc0355@mail4.doshisha.ac.jp), Yoshiki Nagatani (Electronics Eng., Kobe City College of Technol., Kobe, Hyogo, Japan), Ko Chiba (School of Biomedical Sci., Nagasaki Univ., Nagasaki, Japan), Isao Mano (Dept. of Elec. and Electronics Eng., Doshisha Univ., Joyo, Japan), and Mami Matsukawa (Dept. of Elec. and Electronics Eng., Doshisha Univ., Kyotanabe, Japan)

Low intensity pulsed ultrasound can shorten the healing time of a bone fracture; however, the present technique does not seem not to consider the complicated ultrasound propagation in the body with bone. In this study, the time reversal technique was used to effectively converge ultrasound in the bone part. A three-dimensional distal radius model was constructed by using a high resolution-peripheral quantitative CT data (spatial resolution; 61 μm). In the simulation using Finite Difference Time Domain method, one cycle of sinusoidal ultrasound wave (1 MHz) with Hanning window was transmitted from a virtual fracture point (1 mm^3). The model was surrounded by water. The propagating ultrasound waves were observed by two ring shape array transducers, coaxially placed along the bone model at the distance of 30 mm from the fracture point. Then, time reversal waves were radiated from the array transducers, considering the phase shifts and amplitudes of the observed waves in the previous simulation. The time reversal waves successfully converged near the initial virtual fracture part, telling the usefulness of the technique. In the radial-tangential plane including the point, the converged area ($-$3dB of maximum stress value) was 42.8 mm^2.

1:15

3pBAb2. Evaluation of wave velocity in bovine meniscus by laser ultrasound technique. Takumi Fukunaga (Elec. and Electron. Eng., Doshisha Univ., 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, ctwc0309@mail4.doshisha.ac.jp), Masaki Kuraoka, Taiki Makino (Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Mami Matsukawa (Elec. and Electron. Eng., Doshisha Univ., Kyotanabe, Japan)

Laser ultrasound technique is suitable to measure the ultrasound velocity of small samples such as meniscus. In this study, longitudinal wave velocities in bovine meniscus samples were measured using a short pulsed laser. The samples were obtained from the knee joint of bovine femurs and five cubes ($3\times3\times3$ mm^3) were manually manufactured. The ultrasonic wave generated by the laser was propagated through the samples and the corresponding velocities in the axial, radial, and tangential directions were measured. The obtained velocity range was 2820–3560 m/s. The highest velocities were found in the tangential direction, followed by the velocities in the radial and axial directions, except for the samples extracted from the most posterior part of the meniscus, where the velocities were highest in the radial direction. In order to investigate the anisotropy of the meniscus, the cube samples were processed into octagonal prisms, and the velocities in the directions between tangential and axial directions were measured. The velocities observed in these directions were between the values of the ones in the tangential and the axial directions, showing the uniaxial anisotropy.

Invited Papers

1:30

3pBAb3. *In vivo* ultrasonic evaluation of distal part of radius in their early teens. Mami Matsukawa (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Isao Mano, Kaoru Horii (OYO Electric Co., Ltd., Joyo, Japan), Shiori Umemura, and Etsuko Ozaki (Kyoto Prefectural Univ. of Medicine, Kyoto, Japan)

For the evaluation of small radius of early teens, we have improved the ultrasonic bone measurement system, LD-100 (OYO electric)[1], which can evaluate 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. We can also obtain the spatial distribution of wave attenuation during propagation in bone, which can be presented as an attenuation map. In our present teen's bone evaluation project (ages 12-18, n = 971), we found an open epiphysis at the distal end of radius in some attenuation maps of early teens. The area seems to be the growth plate, which have the ability to lengthen the bone by creating new cartilage within the bone itself. In case of boys, the plates were found

at the age of 14 or less. The values of body weights and heights of the teens with the growth plates were smaller than the average, telling that they are growing up. These results show the possibility of growth evaluation using ultrasound techniques. 1. H. Sai, *et al.*, Osteoporos Int. (2010) 21:1781–1790.

1:50

3pBAb4. Ultrasonic characterization of porous gyroid scaffolds for bone tissue engineering: Mechanical modelling and numerical validation. Giuseppe Rosi (MSME, Université Paris-Est, 61 Ave. du Géneral de Gaulle, Créteil 94010, France, giuseppe.rosi@univ-paris-est.fr), Vu-Hieu Nguyen (MSME, Université Paris-Est, Creteil, France), Adrien Loseille (GAMMA3 Team, INRIA, Saclay, France), and Salah Naili (MSME, Université Paris-Est, Créteil, France)

Bone substitutes can be used for pre-implant surgery in presence of volumetric bone defects. In this context, the ultrasound characterization of the bone substitute is a key issue. To this end, we model the implant as a 3D porous structure and we study its ultrasonic behavior. In the framework of artificial bone substitutes, microstructured scaffolds are widely used. In the literature, several geometrical configurations have been tested: among them, the gyroid-shaped scaffolds turn out to be an excellent choice, thanks to its ability to reproduce the behavior and the porous structure of trabecular bone. This study is focused on the mechanical modelling and numerical validation of wave propagation in a porous implant substitute. In particular, 3D finite-difference time-domain (FDTD) simulations were performed on a gyroid-shaped scaffold of saturated with water to validate the continuum mechanical model. Ultrasound excitations at different central frequencies were used in order to investigate the frequency-dependent behavior phase and group velocities.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 ESQUIMALT (VCC), 1:00 P.M. TO 2:25 P.M.

Session 3pEA

Engineering Acoustics and Noise: Acoustic Particle Velocity Sensors, Algorithms, and Applications in Air

Michael V. Scanlon, Chair
RDRL-SES-P, Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Paper

1:00

3pEA1. Airborne vector sensor experiments within an anechoic chamber. 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 (Acoust. and Mech. Eng., Appl. Phys. Lab., Univ. of Washington, Seattle, WA), W. C. Kirkpatrick Alberts, and Michael V. Scanlon (Acoust., Army Res. Lab, Adelphi, MD)

A technique to measure acoustic particle velocity in air is demonstrated in a controlled set of experiments conducted within an anechoic chamber at the Army Research Lab (ARL-Adelphi). This air-borne vector sensor measures acoustic pressure with an omnidirectional microphone and the 3D acceleration of a lightweight, rigid sphere with an embedded high-sensitivity tri-axial accelerometer. Two experiments are presented to demonstrate fundamental properties of the acoustic vector field: sound directionality, and the relative phase of pressure and particle velocity in the near to far field transition. Directional studies are implemented in the far-field of a speaker producing tones, defined by $kr>>1$, where r is the range from the speaker and k is the acoustic wavenumber. The pressure-velocity phase relationship is examined through the transition $kr << 1$, $kr = 1$, and then $kr >> 1$ by changing frequency and sensor range. For comparison, a tri-axial Microflown sensor simultaneously recorded the acoustic field. As neither sensor measures particle velocity directly, the phase characteristics of sensors depend on the sensing technique and corrections must be made to account for the true phase of particle velocity. Advantages and disadvantages of each method are discussed.

Contributed Papers

1:25

3pEA2. Localizing multiple targets with a single particle velocity sensor. Tung-Duong Tran-Luu and Minh Dao (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, tung-duong.tran-luu.civ@mail.mil)

The acoustic velocity sensor (AVS) measures the direction of a propagating plane wave by measuring instantaneously the particle velocity over a small volume in space. When multiple sound sources are present, the AVS no longer reports the correct directions as the sound intensity vectors (the product of pressure and velocity vector) get mixed up nonlinearly. A few methods have been proposed to extract the individual components from just one sensor by assuming, for example, spectral separability or nongaussianity. This paper shows it is in fact possible to extract the components without making such assumptions. The approach here is to constrain the signal with the acoustic impedance equation for multiple superposing plane waves and, applying various techniques, such as sparsity decomposition, to solve the derived least squares problem.

1:40

3pEA3. Seismo-acoustic detection and tracking of small aircrafts. Joydeep Bhattacharyya, Christian G. Reiff, Leng Sim, and W. C. Kirkpatrick Alberts (Sensors and Electronics Directorate, Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, joydeep.bhattacharyya.civ@mail.mil)

The Army Research Laboratory is undertaking an investigation to combine acoustic and seismic vector sensors, in conjunction with traditional acoustic arrays which use omnidirectional microphones, to detect and track small airplanes. The focus of this study is to improve the fidelity of the track with a sparse network and explore its applicability in both urban and rural settings. Towards this end, we carry out a field test where multiple aircrafts with known flightpaths are acoustically and seismically monitored over a period of several days. Using a detection algorithm that leverages both the blade frequency of the propellers and the Doppler associated with near-source propagation, we demonstrate that both acoustic and seismic data have the ability to detect and track the aircrafts. Both acoustic and seismic vector sensors have the ability to estimate both the arrival time and the azimuth to the source which lead to smaller sensor footprint. By combining multimodal sensing with novel target tracking techniques, we can reduce the impact of clutter while maintaining a robustness of our detection and tracking algorithms. A comparison of the acoustic and seismic vector sensors in tracking small aircrafts will be discussed.

1:55

3pEA4. Estimation of the source distance using the multiple three-dimensional acoustic intensimetry. In-Jee Jung and Jeong-Guon Ih (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yusung-gu, Daejeon 34141, South Korea, injee@kaist.ac.kr)

A 3D acoustic intensimetry (AI) in tetrahedral configuration can be used for the source localization. Such a 3D AI module is compact, but it can also yield a precise bearing angle for the low Helmholtz numbers if some bias errors are compensated. In this study, besides the bearing angle, the source distance is estimated by using the multiple 3D AI probes with different acoustic centers. Because each AI module indicates a vector from the acoustic center to the source direction, the nearest point between any pair of line vectors can approximate the source position. The source position can be estimated by minimizing the least square error of the positions determined by multiple intensity vectors. Such idea is preliminarily tested with four 3D AI modules, of which each module is composed of 4 MEMS microphones separated by 30 mm. Four 3D AI modules are also arranged in a tetrahedron layout, and the source frequency range is 0.5–2 kHz. The distance between modules is varied for 150–250 mm range, and the source distance for 1–4 m range. Current test results reveal that the estimation error is less than about 2, 5, and 7% when the elevation angle is within 0–30°, 30–60°, 60–90°, respectively.

2:10

3pEA5. Source localization using back propagation neural network and a single vector receiver. Wenbo Wang, Lin Su, Li Ma, and Qunyan Ren (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renqunyan@mail.ioa.ac.cn)

Matched field processing (MFP) has been widely used in source localization in shallow waters, whose performance is strongly correlated with the knowledge of environmental properties and proper selection of sound propagation model. This paper presents a neural network based approach for source ranging of moving target, which does not need heavy sound field calculation and no requirement of environmental information as a known prior. This neural network is designed to determine source range by observing multiple-frequency sound fields as excited by the source and recorded on a single vector receiver. In synthetic tests, this neural network is first trained on training data set as to adaptively select the optimal features through error back propagation, and then its localization performance is validated on testing data set. This approach is then tested on ship noise data as collected by a single vector sensor, and the relative error is smaller than 0.1 through comparing to GPS calculations. The promising results suggest the proposed approach can be further developed for source ranging applications with light system.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018

PREFUNCTION AREA 1B (VCC), 1:00 P.M. TO 2:00 P.M.

Session 3pEDa

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 2:00 p.m.

3pEDa1. Nonlinear tuning curve vibration using a column of non-wetted or wetted glass beads vibrating over a clamped elastic plate: Case for air-borne sound excitation. Emily V. Santos (Biological Sci., Univ. of Maryland Baltimore County, Phys. Dept., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

A soil plate oscillator (SPO) apparatus consists of two circular flanges sandwiching and clamping a thin circular elastic plate. The apparatus can model certain aspects of the nonlinear acoustic landmine detection problem which involves the interaction of granular material with an elastic plate in flexural vibration. Uniform spherical glass beads—representing a nonlinear mesoscopic elastic material—are supported at the bottom by the acrylic plate (4.5 inch diam, 1/8 inch thick) and stiff cylindrical sidewalls of the upper flange. Two 3 in. diam loud speakers placed above the bead column are driven with a swept sinusoidal signal applied to a constant current amplifier to generate airborne sound excitation from 50 to 1250 Hz. A small accelerometer fastened to the underside of the plate at the center measures the response. Separate tuning curve experiments are performed using a fixed column of 350 grams using 2,3,…,10 mm diameter soda-lime glass beads—wetted or non-wetted (mineral-oil). With dry beads, backbone curves (peak acceleration vs. corresponding resonant frequency) exhibit a linear region with comparable slopes, while detailed back-bone curvature vs. bead diameter reveals more structure. Wet beads exhibit less non-linear tuning curve behavior.

3pEDa2. Experiments in linear and nonlinear acoustic landmine detection. Francesca M. Browne and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m200726@usna.edu)

When nonmetallic land mines are difficult to detect using ground penetrating radar, acoustic landmine detection is used as a confirmation sensor. Non-metallic mines include VS 1.6 and VS 2.2 anti-tank plastic landmines. In modeling nonlinear acoustic landmine detection, a 20 cm diam drum-like landmine simulant (5.08 cm height, 0.64 cm thick aluminum bottom, 0.64 cm thick acrylic top plate) was used. The flexural vibration of the acrylic top plate with the soil behaves nonlinearly. A concrete soil tank filled with dry sifted masonry sand served as an idealized granular medium. The soil tank was $57.2 \times 57.2 \times 22.9$ cm deep (wall thickness 15.2 cm). The simulant is 3 cm deep. In airborne excitation experiments, two 8 inch diam subwoofers radiate a swept tone. A geophone (1 inch diam) measured the rms particle velocity at 28 scan locations across a 56 cm segment on the surface. The geophone-microphone frequency response (between 50 Hz and 450 Hz) vs. scan position is used to extract a response profile at any individual frequency. The lowest drum-like mode shape was strongest at 156 Hz. Nonlinear tuning curve measurements at fixed locations exhibit more softening “on the mine” vs. “off the mine.”

3pEDa3. Demonstration of acoustic landmine detection using a clamped soil plate oscillator with airborne sound excitation. Jenna M. Cartron (Phys. Dept., U.S. Naval Acad., 3519 Forest Haven Dr., Laurel, MD 20724, Jcartron11@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

Landmine detection in gravel road beds, soil, or sand for some mines (constructed out of plastic) are difficult to detect using ground penetrating radar. In acoustic landmine detection, one would like to remotely detect a vibration profile across the surface of the soil (sand or gravel) due to the interaction between a large soil column and the localized elastic top plate. In an idealized laboratory experiment, the soil plate oscillator (SPO) apparatus models an idealization of an acoustic landmine detection experiment. The clamped circular acrylic plate (1/8 in. thick) models the top plate of a plastic buried mine while the soil column (350 g dry sifted masonry sand) supported by the plate models the burial depth. A constant current amplified sinusoidal chirp from a sweep spectrum analyzer drives two three inch diam speakers (placed above the column) generating airborne excitation. A laser Doppler vibrometer (LDV) measured the rms particle velocity at 15 scan locations across the 4.5 inch diameter soil surface in sweeps between 50 Hz and 1250 Hz. The LDV-microphone frequency response vs. scan position is used to extract a response profile at any individual frequency. These mixed mode shapes are compared for both in-phase and out-of-phase excitation.

3pEDa4. Characteristics of the soundscape before and after the construction of the Block Island Wind Farm. Aditi Tripathy (Ocean Eng., Univ. of Rhode Island, Union Express, 50 Lower College Rd., Unit 4700, Kingston, RI 02881, atripathy@my.uri.edu), James H Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer Amaral, Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Adam S. Frankel (Marine Acoust., Inc., Arlington, VA), Ying-Tsong Lin, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Block Island Wind Farm (BIWF) south of Rhode Island is the first offshore windfarm in the United States. As part of the Ocean Special Area Management Plan, acoustic data were collected before the construction in the fall of 2009. Noise budgets were estimated based on this data and showed the dominant sources of sound in a 1/3-octave band centered at 500 Hz were shipping and wind. Data were again collected during and after construction of the wind farm and will be presented and compared to pre-construction levels. In 2009, Passive Aquatic Listener (PALs) were deployed. After construction was complete, data from a tetrahedral hydrophone array (~50 m from one of the wind turbines) were analyzed to study the soundscape from December 20, 2016 to January 14, 2017. The acoustic environment near the BIWF after construction showed contributions from shipping, wind, and marine life. Noise from the wind turbine was measured near 70 Hz at approximately 100 dB re 1 mPa at a range of 50 m. Significant marine mammal vocalizations were recorded including from humpback and fin whales. (Work supported by the Bureau of Ocean Energy Management.)

3pEDa5. Speech-language pathology student–clinicians’ self-awareness of tongue position during rhotic sound production in American English. Megan Diekhoff and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Proprioceptive signals from tongue positions are not strong, meaning it can be challenging to teach the articulation of speech sounds to language learners. Some of the most prevalent speech sound disorders in American English involve the rhotic /ɹ/. Speech-language pathologists provide modeling and feedback to their clients during treatment. Since tongue shape is not readily visible during rhotic production, clinicians rely heavily on verbal descriptions and auditory feedback to explain sounds to their clients. It is currently unclear how closely these approaches mirror or depend on veridical descriptions of tongue shape. An example from musical education provides an instructive backdrop: Although clarinet teachers frequently describe raising the tongue body for high notes, the tongue body must physically lower [Lulich *et al.*, 2017 JASA]. Students successfully learn to play high notes regardless of the accuracy of the description, but would clarinet instruction be more effective if it reflected physical reality? Analogous issues and questions can be raised in the context of speech-language pathology. This study presents preliminary results of an investigation of student-clinicians’ self-awareness of tongue position during rhotic sounds.

Session 3pEDb

Education in Acoustics: General Topics in Acoustics Education

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Contributed Papers

2:30

3pEDb1. An Acoustical STEAM event for high school students. David A. Lechner (Mech. Eng., The Catholic Univ. of America, 9404 Bethany Pl., Washington, DC 20064, 66Lechner@cua.edu), Otto Wilson (Biomedical Eng., The Catholic Univ. of America, Washington, DC), and Shane Guan (Mech. Eng., The Catholic Univ. of America, Silver Spring, MD)

This presentation will review the methods and results of hosting our first Acoustical STEAM (Science, Technology, Engineering, Arts, and Math) competition as a part of the Washington DC Regional Chapter meeting of the ASA in the spring of 2018. Student teams were invited from over 50 Washington DC High School clubs and physics classes, provided a box assorted materials, and given 2 hours to create a musical video. The presentation will discuss the objectives, lessons learned, and results of the competition as well as student feedback on the event.

2:45

3pEDb2. A measurements laboratory built around a digitally-sampled microphone circuit. Andrew B. Wright (Systems Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave., EIT 522, Little Rock, AR 72204, abwright@ualr.edu), Ann Wright (Phys., Hendrix College, Conway, AR), and Amanda Nolen (School of Education, Univ. of Arkansas at Little Rock, Little Rock, AR)

In Fall of 2017, a series of measurement laboratory exercises, built around a microphone preamplifier and filter, were used in teaching an engineering measurement techniques course. The microphone circuit included a preamplifier that introduced JFETs, a virtual ground and buffer to illustrate loading effects, a high pass filter and a low pass filter, and a zener-diode-based limiter. The data were sampled using a beaglebone microcontroller's built-in analog-to-digital converter. The data was transmitted to matlab for post-processing and analysis. The microphone was then used in the Spring 2018 acoustics class. A variety of projects, including a Kundt's tube, a driven closed-open pipe, a 3D printed horn for a blue-tooth speaker, and an absorptivity measurement for a restaurant's wall treatment tiles, were completed in the course. It was observed that students showed improved interest in the material (increased in-class and out-of-class questions about specifics of the course material related to the project, increased exploration of material outside the provided course materials, full participation by all members of the class in the project) and learning (improved test scores after the project began relative to scores before the project began) relative to earlier classes where no experimental component was included.

3:00

3pEDb3. Vibra-Son: A general audience exhibition dedicated to acoustics in Sherbrooke, Québec. Olivier Robin (Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Olivier.Robin@USherbrooke.ca)

This presentation will present the ongoing setup of a general audience exhibition dedicated to acoustics in Sherbrooke, Québec. This project is built up with the Sherbrooke Nature and Science museum, Sherbrooke University (faculty and research groups) and with the contribution of external structures (like the Canadian acoustical association and companies). The exhibition will begin in 2019, June. The design of experiments that allows highlighting the interdisciplinary nature of acoustics is first described, as well as their setup in a limited space including possible travelling of the exhibition. The development or use of mobile applications to help sharing data or results with the visitors is especially under focus. The question of how including research projects into the exhibition is then discussed, with a dual objective of raising awareness of visitors about research but also making them a part of a research operation. Finally, satellites events that are currently planned are described, as well as their role in providing a multimodal description of acoustics, like a reading tour of comics in a library.

3:15

3pEDb4. Mathematica simulation of circular synthetic aperture acoustic imaging. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

A Mathematica simulation will simulate circular synthetic aperture radar (CSAR) and its acoustic counterpart (CSAA). Two-dimensional and three-dimensional point targets will be imaged from echoes collected by a co-located transmitter/receiver moving along a circular track above the target plane. Previous work involved a Mathematica® (ver. 11) simulation of Synthetic Aperture Sonar (SAS). This simulation demonstrated how two-dimensional and three-dimensional point targets on a ground plane can be imaged from a collection of acoustic echoes using stripped-map geometry or a straight line track with a co-located transmitter/receiver element. The back-projection algorithm is used in both imaging geometries to construct a reflectance image of positioned point targets. The upcoming simulation will provide a comparison between circular track geometry and stripped-map straight line geometry.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Subha Maruvada, Chair
U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Information retrieval from a soundscape by using blind source separation and clustering. Tzu-Hao Lin (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., 128 Academia Rd., Section 2, Nankang, Taipei 115, Taiwan, schonkopf@gmail.com), Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan), Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan), Mao-Ning Tuanmu (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan), and Katsunori Fujikura (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., Yokosuka, Japan)

Passive acoustic monitoring represents one of the remote sensing platforms of biodiversity. However, it remains challenging to retrieve meaningful biological information from a large amount of soundscape data when a comprehensive recognition database is not available. To overcome this issue, it is necessary to investigate the basic structure of a soundscape and subsequently retrieve biological information. The recent development of machine learning-based blind source separation techniques allow us to separate biological choruses and non-biological sounds appearing on a long-term spectrogram. After the blind source separation, the temporal-spatial changes of bioacoustic activities can be efficiently investigated by using a clustering algorithm. In this presentation, we will demonstrate the information retrieval in the forest and marine soundscapes. The separation result shows that in addition to biological information, we can also extract information relevant to weather patterns and human activities. Furthermore, the clustering result can be used to establish an audio library of nature soundscapes, which may facilitate the investigation of interactions among wildlife, climate change, and human development. In the future, the soundscape-based ecosystem monitoring will be feasible if we can integrate the soundscape information retrieval in a large-scale soundscape monitoring network.

1:25

3pID2. Yanny or Laurel? Acoustic and non-acoustic cues that influence speech perception. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61822, monson@illinois.edu)

"What do you hear?" This question that divided the masses highlights the complex nature of speech perception, which is dependent upon both acoustic and non-acoustic information available to the brain. In this talk, I will discuss some of the most recent advances in our understanding of both types of information that enable the complicated task of speech perception under adverse listening conditions. Recent research from our lab and others has revealed a surprising amount of speech signal information resides in the highest audible frequency range for humans (*i.e.*, beyond 8 kHz), providing localization, phonetic, and other speech source information. These cues, as well as non-acoustic (*e.g.*, experiential, visual, lexical) cues likely become increasingly important when a listener is faced with ambiguity and/or degradation of the acoustic signal. Given the diversity of prior experience from individual to individual, differences in perception of an ambiguous/degraded speech signal are to be expected.

1:45

3pID3. DJ Prof: Mixing instructional modes to improve student learning. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu)

Active learning encompasses a broad variety of activities designed to engage students with material while instructors are present to answer questions. In contrast, traditional lecture instruction is passive, and students' initial interaction with material is relegated to homework, which is often completed alone. Freeman *et al.*'s meta-analysis of 225 studies comparing lecture and active learning concluded that active learning is the "preferred empirically validated teaching practice" and showed that active learning courses have significantly lower failure rates than lecture courses [PNAS, 2014]. Incorporating proven active learning techniques into acoustics courses will improve student learning. In Acoustics Today [2016], John Buck, Jill Nelson, and I compared the professor in an active learning course to a DJ mixing different modes of instruction. DJ Prof designs a student-centered environment combining collaborative in-class exercises, short lecture segments, online video examples, and reading assignments. If you're an active learning skeptic, this talk will highlight empirical evidence of its benefits. If you're interested in getting started with active learning, this talk will suggest easy exercises to try. If you're an experienced DJ Prof, this talk will provide examples to amp up your pedagogical playlist.

Session 3pNS

Noise: Wind Turbine Noise

Kerrie G. Standlee, Chair
Daly-Standlee & Associates, Inc., 4900 SW Griffith Drive, Suite 205, Beaverton, OR 97005

Chair's Introduction—1:00

Contributed Papers

1:05

3pNS1. Simulated wind turbine emissions: Results from additional human response testing. Peggy B. Nelson, Matthew Waggenspack, Andrew Byrne (Dept of Speech-Language-Hearing Sci.,Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Michael Sullivan (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Meredith Adams (Otolaryngol., Univ. of Minnesota, Minneapolis, MN), and Jeffrey D. Marr (St. Anthony Falls Lab., Univ. of Minnesota, Minneapolis, MN)

Previous data (Nelson *et al.*, ASA 2017) indicated that healthy human adult participants experienced few symptoms from re-created wind turbine sound and infrasound emissions. That report included more than fifty subjects ages 21–73 years who attended to audible and infrasound signals generated from a wind turbine, recorded at 300 meters and re-created in a laboratory. Stimuli consisted of modulated and unmodulated audible turbine sound at 50 dB SPL, as well as natural and peak-enhanced turbine infrasound at an overall level of approximately 85 dB SPL (peaks up to 100 dB SPL). Participants were tested for their postural stability, detection, and ratings of audible and infrasound emissions randomly presented in one-minute exposure intervals in the laboratory. Very few and minor adverse effects had been noted to date, mostly ear fullness or pressure. Healthy participants showed no evidence of any change in postural sway in the presence of infrasound for the group tested. We have recently begun testing of participants who either a) live near turbines and complain of adverse effects, or b) who have symptoms of dizziness/imbalance as reported to their ENT specialist. Results from postural sway, sound quality judgments, and pre- and post-exposure symptoms will be reported from some of these participants. [Work supported by Xcel Energy RDF14.]

1:20

3pNS2. Audible noise sources (or characteristics?) from wind turbines using acoustic beam-forming. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

In many investigations of wind turbine noise, residents report specific and varying types of audible characteristics. With respect to the subject audible characteristics reported by residents (which are not picked in the dBA Leq measurements) turbines exhibit multiple audible sources at different operating and environmental conditions. Imaging techniques for real-time noise source identification have been used to identify tonal as well as dynamic pulsation during various environmental conditions and distances. Challenges with these techniques are presented using advanced imaging algorithms with successful imaging.

1:35

3pNS3. Existing aeroacoustic issues of building elements. Michael Bolduc (RWDI Air Inc., 600 Southgate Dr., Guelph, ON N1G 4P6, Canada, Michael.Bolduc@rwdi.com) and Andrew Bell (RWDI Air Inc., Copenhagen, Denmark)

During high wind events, certain building elements have the potential to generate significant tonal noise that can be heard miles away. These aeroacoustic issues are due to an interaction/coupling between an unstable fluid flow and a feedback mechanism, usually being a resonant acoustic mode. Aeroacoustic issues are often easy to avoid when susceptible geometry can be identified during the early design process, but mitigation can be technically difficult and very costly once issues do occur. Because aeroacoustic issues are unique and often annoying, they are often considered newsworthy and can be the subject of significant publicity. A series of case studies of existing structures, which have experienced aeroacoustic problems from their surrounding wind climate, are presented. Through each of these case studies, the steps used to identify suspect features and the recommended remedial measures are presented. The methodology for identification of the suspect feature may include investigating trends in weather data, conducting spectral analysis of audio recordings, performing calculations of the natural frequencies of building features, and on-site observations and measurements. Various mitigation strategies are detailed, depending on the mechanism found to be generating the aeroacoustic noise.

1:50

3pNS4. Wind turbine infra-sound penetration inside homes from multiple variable speed turbines. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Previous measurements in homes with respect to dominant turbines indicated the transmissibility factor calculations were a valid technique for describing wind turbine infra-sound penetration at various locations inside homes. However, when several turbines contribute at different RPM's and/or are not in phase different techniques need to be considered due to RPM variation and FFT densities. A comparison of the different methods is shown with examples of both cases.

Session 3pPP

Psychological and Physiological Acoustics: Pitch and Sound Localization

Ilse B. Labuschagne, Chair
Audiology and Speech Sciences, University of British Columbia, 4478 Haggart St., Vancouver, BC V6L 2H5, Canada

Contributed Papers

1:00

3pPP1. On the uncertainty principle for the pitch of brief tones. William M. Hartmann and Jon Pumplin (Phys. and Astronomy, Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu)

Many experiments have shown that the ability of human listeners to discriminate the frequencies of brief tones can violate the wave uncertainty principle that applies to linear systems. Several observations can be made: (1) Experiments that associate the duration of the tone envelope, or the second central moment of the squared envelope, with temporal uncertainty are consistent with the application of the energy-time principle in non-relativistic quantum mechanics. (2) The role of echoic memory in confounding the definition of duration can be avoided by immediately following the target tone to be judged by a tone of different frequency. (3) Whereas the uncertainty principle relates well-defined conjugate variables, such as time and frequency, listeners in psychoacoustical experiments may benefit from comparisons in both the pitch dimension (coding frequency) and a timbral dimension associated with the spectral distribution of energy. The latter provides additional information that is outside the context for the uncertainty principle. Its influence can be reduced by experiments in which the listener compares target tones having different brief durations. When these considerations are applied, it is found that human listeners still beat the uncertainty principle, but not as impressively.

1:15

3pPP2. Thresholds of noise in harmonic series maskers. Ilse B. Labuschagne and Valter Ciocca (Audiol. and Speech Sci., Univ. of Br. Columbia, 4478 Haggart St., Vancouver, BC V6L 2H5, Canada, ilse.labuschagne@alumni.ubc.ca)

Numerous studies have investigated the detection of pure tones and harmonic series in noise, but far fewer studied the perception of noise in harmonic series. One such study [Gockel, Moore and Patterson, J. Acoust. Soc. Am., **111**(6), 2759–2770 (2002)] demonstrated that the masking thresholds of noise in harmonic series maskers were affected by the fundamental frequency (F0) and the overall level of the series, and by the relative phase of the harmonic components. The maskers used by Gockel *et al.* comprised unresolved harmonics (10^{th} and higher harmonics below 5 kHz) for which F0 and frequency range covaried. The current study investigated noise detection for three non-overlapping spectral bands of equal auditory filter bandwidths (ERBs). Bands included either resolved harmonics (B1), unresolved harmonics (B3), or both resolved and unresolved harmonics (B2). Masker F0 and overall level were also varied. A Bayesian linear mixed-effects analysis showed that noise detection was better for higher F0s in B1 and B2, and that a higher presentation level resulted in a small improvement in noise detection in B2. Noise detection was better for higher presentation levels in B3. The findings will be discussed in relation to predictions of auditory processing models [Patterson, Allerhand, Giguère, J. Acoust. Soc. Am. **98**(4), 1890–1894 (1995)].

1:30

3pPP3. The sound image trajectory of a moving sound approaching a reflective wall. Shunya Kurachi, Daisuke Morikawa, and Tatsuya Hirahara (Dept. of Eng., Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, t754008@st.pu-toyama.ac.jp)

The movement of a sound image may not coincide with that of the sound source in a room with reflective walls. A listener sitting by a reflective wall perceives the movement of the sound image bending around their head when a sound source approaches the wall straight in the front of them. In order to investigate this perception quantitatively, we measured the trajectory of the sound image perceived by a listener against a sound source moving straight in front of them using a magnetic motion sensor. The sound source was a 1/3-octave band noise with a center frequency of 300, 750, 1500, 3000, or 6000 Hz emanating from a small moving loudspeaker at a sound pressure level of 70 dB. The loudspeaker moved on a rail perpendicular to the wall in front of the listener at a constant linear velocity. The listener tracked the direction of the moving sound image with his fingers by holding a motion sensor with his eyes closed. Results show that the sound image trajectory bent around the listener's head only when the stimulus contained low frequency components below 1.5 kHz, suggesting that interaural time difference plays an important role in the phenomenon.

1:45

3pPP4. An investigation of the training effects in a seven-week auditory localization training using augmented reality. Song Hui Chon and Sungyoung Kim (RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com)

We present the results from a seven-week auditory localization training using augmented reality with Microsoft HoloLens. For each trial, the target and distractor(s) would be randomly placed in space. In training, after the participant estimated the invisible target location, both true and estimated locations would be visually displayed. A localization score was calculated and recorded based on their distance. The test modules were identical to the training modules, except that the true location was not shown. The target was a female singing, and the distractor a piano accompaniment. Eight listeners were divided into two groups after the initial test, keeping the mean and variance of localization scores at the same level. Both groups showed a declining pattern in localization score over the eight tests, unlike the results from our previous study. The decreasing slope was smaller for the train group than the control group, which might reflect some mild effect of training. There was a one-time performance improvement after two trainings. The experiment might have been too simple to maintain participants' attention for weeks. Possible extraneous factors such as academic schedule might also have had an impact on this decreasing pattern.

2:00–2:15 Break

2:15

3pPP5. Adapting masking thresholds for spatially separated sounds in two dimensional ambisonics. Yuval Adler and Prateek Murgai (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct, Stanford, CA 94305, adler@ccrma.stanford.edu)

Spatial audio reproduction has gained renewed interest in recent years with the increasing presence of virtual reality applications. Ambisonics is currently the most widely adopted format for spatial audio content distributed via internet streaming services, and this has raised the need for audio compression relevant to the format. Current perceptual audio coders used for stereo audio content rely heavily on masking thresholds to reduce data rates, but these thresholds do not take into consideration spatial release from masking. This study begins an effort to update these thresholds for spatially separated sources in ambisonics. The listening tests were performed with sounds encoded in ambisonics to allow for tests to integrate whatever inherent limitations exist in the format. Initial listening tests were carried out for a subset of possible conditions—sound sources were separated along the horizontal plane, with a specific set of separation angles between the masker and maskee. Suggestions are given for continuing the work for the full range of possible conditions.

2:30

3pPP6. Right/left asymmetricity of horizontal sound localization of younger people. Tsukuru Osawa, Kazumoto Morita, Kenta Toyoda (Precision Mech. Eng., Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, tsukuru_osawa@camal.mech.chuo-u.ac.jp), Jo Sakashita, and Takeshi Toi (Precision Mech. Eng., Chuo Univ., Bunkyo-ku, 121-8551, Japan)

To investigate the performance of horizontal sound localization of younger people, authors conducted a test using 12 young participants (11 males, 1 females; ages 21–24.) The stimuli were presented through headphones to their right and left ears with 4 interaural time differences (ITDs) (0.2, 0.4, 0.6, and 0.8 ms) and 4 interaural level differences (ILDs) (3, 6, 9, and 12 dB(A)) besides the same condition to both ears. Concerning the frequency, pure tones of 0.5, 1.0, and 2.0 kHz, and composite sound of 0.5 + 2.0 kHz were presented, which lasted one second for each trial. The participants answered "Right" or "Left" after each trial according to their own judgment. The results of ITD condition showed that a right-sided inferiority was recognized in case of 1.0 and 2.0 kHz. Concerning the ILD condition, no right/left asymmetricity was shown. This asymmetricity may be caused by the improper setting of the experiment apparatus. Therefore, we switched right/left of the headphones to carry out an additional test in the same way. The same tendency as the original test was obtained, which means that the asymmetricity is likely due to the native characteristics of the participants.

2:45

3pPP7. Characteristics of horizontal sound localization of elderly people and analysis of its potential influential factors. Kazumoto Morita, Tsukuru Osawa, Kenta Toyoda, Jo Sakashita, and Takeshi Toi (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com)

To investigate the performance of horizontal sound localization of elderly people, authors conducted a test using 19 elderly participants (10 males, 9 females; ages 65-85; mean 71.5 years old.) The stimuli were presented through headphones to their right and left ears with 4 interaural time differences (ITDs) (0.2, 0.4, 0.6, and 0.8 ms) and 4 interaural level differences (ILDs) (3, 6, 9, and 12 dB(A)) besides the same condition to both ears. Concerning the frequency, pure tones of 1.0 and 2.0 kHz, and composite sound of 0.5 + 2.0 kHz were presented, which lasted one second for each trial. The participants answered "Right" or "Left" after each trial according to their own judgment. The results of ITD condition showed that in case of the pure tones a right-sided inferiority was recognized, which was poorer than the results of the younger participants. Meanwhile, no right/left difference was recognized with the composite sound. Concerning the ILD condition, no right/left asymmetricity was shown. Furthermore, no significant differences were recognized regarding the relationships between their sound localization performance and potential influential factors. The factors were Trail Making Test (TMT) results, their preferred ear, and their hearing loss difference between right/left ears.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 SAANICH 1/2 (VCC), 1:00 P.M. TO 2:50 P.M.

Session 3pSA

Structural Acoustics and Vibration, Noise, and Signal Processing in Acoustics: History of Computational Methods in Structural Acoustics and Vibration

Micah R. Shepherd, Cochair
Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

Trevor W. Jerome, Cochair
Department of Acoustics, The Pennsylvania State University, GTWT Water Tunnel, State College, PA 16804

Invited Papers

1:00

3pSA1. Historical development of damping models in structural acoustics and vibrations. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

This presentation discusses historical developments of damping models most commonly used in the analysis of structural acoustics and vibrations today. This presentation represents all damping models as frequency-dependent complex-valued matrices that appear in the equations of motion. The historical development of damping models has benefited from research on two fronts. The first is satisfaction of passivity and causality conditions by the damping model. The passivity condition requires no net power flow from the material over a cycle of vibration, while the causality condition requires that the material cannot respond before the excitation begins. The passivity condition is clearly stated in the frequency domain but is less obvious in the time domain. Conversely, the causality condition is clearly stated in the time domain but less obvious in the frequency domain. The second front is the affect of each damping model on computational methods used to evaluate the equations of motion. Discussion will include the use of undamped eigenvectors and Krylov spaces to create reduced-order models. Examples involving a rectangular plate will be presented to illustrate the features of commonly used damping model.

1:20

3pSA2. Determining acoustic sound power using vibration measurements and acoustic radiation modes. Caleb B. Goates (Dept. of Phys., Brigham Young Univ., Provo, UT), Cameron B. Jones (Mechanical Eng., Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu), and Jonathan Blotter (Mechanical Eng., Brigham Young Univ., Provo, UT)

In this paper, an efficient method for measuring the radiated sound power from consumer, industrial, and military products is presented. The method is based on spatially-dense vibration measurements from a 3D laser vibrometer and the acoustic radiation modes approach for computing sound power. The method will be a design tool that will allow engineers and designers to design quieter products by more effectively using noise as a design constraint. There are several standards that describe methods to compute sound power but these methods typically require specialized acoustic chambers for high accuracy results. The method presented here can be performed in an unknown acoustic environment without a reflective surface or an array of specific measurement points surrounding or half surrounding the noise source. The method will also be insensitive to varying background noise, temperature, and fluid/wind flow. The method will provide near real-time results and will be capable of measuring the radiated sound power from complex 3D geometries and built-up structures or parts of these structures. The method will presented in detail with experimental results validated the process.

1:40

3pSA3. Structural intensity on shell structures via a Finite-Element-Method approximation. Felipe Pires, Steve Vanlanduit, and Joris Dirckx (Biophys. & Biomedical Phys., Univ. of Antwerp, Groenenborgerlaan 171, Antwerp 2020, Belgium, felipe.pires@uantwerpen.be)

The study of energy transmission on plate-like structures is widely documented in literature. Its analysis requires the computation of the spatial derivatives of the out-of-plane displacement field of the studied sample and a priori knowledge of its material properties. However, if the structural intensity is to be assessed on irregular shells, such a study requires a more elaborate data processing. In addition to in-plane displacements, also the spatial derivatives along the sample's local coordinates are needed. For this purpose, a method was developed to approximate both the experimental displacement data and the spatial coordinates of a given arbitrary shell using a Finite-Element-Method model. After measuring the global displacement fields and their corresponding spatial coordinates for a given sample, the data was transferred to a shell model whose basis functions were properly defined in accordance to the application's demands and whose individual elements were assumed to behave in accordance with the Kirchhoff plate theory. The proposed method was able to process the experimental displacement and shape data of shells structures and proved to be a reliable tool to assess energy transmission.

2:00

3pSA4. A quasi-analytical formulation for acoustic radiation modes of simple curved structures. Caleb B. Goates, Scott D. Sommerfeldt, and Jonathan Blotter (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com)

Acoustic radiation modes have become a useful and widespread analysis tool in situations involving sound radiation from vibrating structures. They have found use in applications such as active structural acoustic control, optimization of structures for minimal sound radiation, and acoustical holography. Analytical expressions for the radiation resistance matrix, from which the radiation modes are obtained, are available only for a small number of simple source geometries, while the obtaining of radiation modes for more complicated structures typically requires BEM or similar computational methods. This paper details the development of quasi-analytical expressions for the radiation resistance matrices of singly-curved structures such as curved plates, cylinders, and angularly truncated cylinders. It is shown that the radiation modes of these structures may be obtained with a relatively low computational load compared to conventional methods.

Contributed Papers

2:20

3pSA5. On the physical meaning of subsonic wavenumber truncation as applied to source identification on vibration structures. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu), Scott D. Sommerfeldt, and Michael T. Rose (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Wavenumber methods have been used to identify the supersonic portions of a vibrating structure that radiate to the farfield. To estimate the supersonic wave energy of a vibrating structure, the discrete Fourier transform can be used to determine the wavenumber spectrum which is then truncated above the acoustic wavenumber. The purely supersonic wavenumber spectrum can now be transformed back into the spatial domain to determine the vibration pattern associated only with the supersonic waves. Often, a cut-off coefficient associated with the acoustic wavenumber of the spatial radiation filter is used to reduce error. Equivalent spatial convolutions have also been formulated to obtain supersonic components of a vibrating pattern. This paper discusses the physical meaning of wavenumber truncation and its accuracy in identifying the surface areas of a vibrating structure that radiate sound. It is shown that truncating subsonic components of the vibration pattern will create non-physical vibration patterns which lie outside of the physical boundaries of the structure. Thus, subsonic wavenumber truncation methods may not be a reliable method for determining the radiated portions of a vibration pattern.

2:35

3pSA6. Experimental and numerical techniques for lightweight vehicle sound package development. Yuksel Gur (Res. and Adv. Eng., Ford Motor Co., Ford Res. and Innovation Ctr. (RIC), 2101 Village Rd., MD3135, Dearborn, MI 48121, ygur@ford.com)

The use of lightweight structural materials poses great challenge to noise control of a vehicle. A new approach to sound package design in lightweight vehicles was developed to reduce vehicle interior noise level without addition of weight. This new approach relies on lightweight acoustical materials that provide superior sound absorption performance to reduce noise level at the source, e.g., inside engine compartment and near the tires. Vehicle NVH finite element analysis (FEA) models were used to develop the lightweight vehicle's structural design to bring the lightweight vehicle's low and mid frequency responses closer the baseline vehicle's performance. Full vehicle SEA (statistical energy analysis) model was developed to evaluate the high frequency NVH (noise, vibration, and harshness) performance of a vehicle. This correlated SEA model was used for the vehicle sound package optimization studies to develop sound package design to improve lightweight vehicle's acoustic performance. In this paper, the use of NVH CAE (computer aided engineering) simulations for lightweight vehicle design is presented to highlight the limitation and use of FEA and SEA in lightweight vehicle development.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 UPPER PAVILION (VCC), 1:00 P.M. TO 3:30 P.M.

Session 3pSC

Speech Communication: Second Language Speakers and Listeners (Poster Session)

Nadya Pincus, Chair
Linguistics and Cognitive Science, University of Delaware, 812 Lehigh Rd., Newark, DE 19711

All posters will be on display from 1:00 p.m. to 3:30 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 2:15 p.m. and authors of even-numbered papers will be at their posters from 2:15 p.m. to 3:30 p.m.

Contributed Papers

3pSC1. English phoneme recognition of vocoded speech by adult Mandarin-speaking English-learners. Jing Yang (Dept. of Commun. Sci. and Disord., Univ. of Wisconsin-Milwaukee, Enderis Hall, 2400 E. Hartford Ave., Milwaukee, WI 53211, phial2000@gmail.com) and Li Xu (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

The present study examined the joint influence of listeners' language experience and the degree of spectral degradation of speech signals on English phoneme recognition of L2 listeners. The participants included 27 native English-speaking listeners and 43 native Mandarin-speaking listeners who used English as an L2. The L2 participants varied in chronological age, age of English learning, length of residency in the United States, and the amount of daily-based English usage. The speech stimuli included 12 English vowels embedded in a /hVd/ context produced by four speakers and 20 English consonants embedded in a /Ca/ context produced by two speakers. The speech stimuli were processed using 2-, 4-, 6-, 8-, and 12-channel noise vocoders. The processed and original stimuli were presented to the listeners for identification in a random order. The results showed that spectral degradation had more adverse effects on the L2 listeners but the L2 disadvantage became more evident as the number of frequency increased. The L2 listeners showed different confusion patterns from the L1 listeners, which was affected by the L2 listeners' native language experience. Furthermore, the regression analysis revealed that the L2 listener's length of residency in the United States was a significant predictor for their phoneme recognition outcomes.

3pSC2. The lexical representation of second language length contrasts: Native English speakers learning Japanese. Rachel Hayes-Harb and Shannon L. Barrios (Univ. of Utah, 255 S. Central Campus Dr. Rm 2300, Salt Lake City, UT 84112, r.hayes-harb@utah.edu)

Adults experience difficulty using novel second language (L2) phonological contrasts to distinguish words. Indeed, even the ability to perceive and/or produce a novel contrast with relative accuracy does not guarantee an ability to implement the contrast to distinguish words in tasks that require lexical access. These observations lead to questions regarding the phonological content of learners' lexical representations of difficult L2 contrasts. We

employed an artificial lexicon study to examine the lexical encoding and implementation of Japanese consonant and vowel length contrasts by native English speakers. In the first experiment, native English speakers were taught a set of six Japanese-like auditory minimal pairs with pictured meanings. The members of each pair were differentiated by vowel length (e.g., [teki] vs. [teeki]). Participants were then asked to match the pictures to auditory words in four conditions: matched (see picture of 'teki', hear [teki]), vowel length mismatch (see 'teki', hear [teeki], and consonant length mismatch (see 'teki', hear [tekki]). Participants performed accurately on matched items, but were more likely to reject word forms mismatched for consonant than for vowel length. The results from this and subsequent experiments provide insight into the lexical encoding strategies used by learners for difficult novel contrasts.

3pSC3. Non-native perception of regionally accented speech of competing talkers. Ewa Jacewicz, Sasha Kim, 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)

Noisy conditions are challenging to non-native listeners who typically underperform relative to native listeners when attending to several competing talkers. We examine whether non-native listeners can utilize dialect-related cues in the target and in the masking speech, even if they do not reach the proficiency level of the native listeners. Our previous work with highly proficient Indonesian-English bilinguals (Fox *et al.*, 2014) found that their performance differed markedly from native English listeners when the speech levels of the competing talkers were equal (0 dB SNR). We hypothesized that bilinguals cannot effectively separate utterances at 0 dB SNR due to a lack of a sufficient contrast in the voice levels of competing talkers, which may reduce their ability to benefit from the phonetic-acoustic details in order to "follow" a particular talker. The current study sought to replicate and validate earlier findings with a different group of bilinguals. The same experiment was conducted with Korean-English bilinguals. The results were replicated, providing further evidence that the ability to benefit from fine-grained phonetic details in regional accents declines for non-native listeners when the speech levels of the competing talkers are equal. Discussion will focus on defining the nature of the non-native speech processing deficit.

3pSC4. Effects of formant proximity and language experience on subcortical neural encoding of vowels in adulthood. Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, University of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu), Matthew Masapollo (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), and Lucie Menard (Dept. of Linguist, Univ. of Quebec at Montreal, Montreal, QC, Canada)

Formant frequency convergence (or "focalization") and linguistic experience interact to shape the perception of vowels in adulthood. Here, we provide evidence of the effects of formant proximities and language experience at subcortical levels of the auditory pathway. Using a passive oddball/reversed oddball paradigm, the frequency-following response (FFR) in the auditory brainstem was elicited in sixteen healthy monolingual English speakers by a less-focal/English prototypic /u/ and a more-focal/French prototypic /u/. We examined the FFR as a function of stimulus type (English vs. French prototype) and condition (Standard vs. Deviant). A cross-correlation analysis revealed higher overall similarity between the FFR and evoking stimulus for the English prototype than the French prototype, suggesting an effect of language experience. Yet, deviants exhibited higher correlational values than standards. This effect was largely driven by the French prototype, suggesting an influence of focalization. A subsequent Fourier analysis revealed an effect of stimulus type at the *f0* peak, whereas only a condition effect was observed at the first harmonic peak. The latter effect was largely driven by the French prototype. Altogether, these findings suggest that subcortical encoding of vowels in adulthood is influenced by a complex interplay between formant proximities and long-term linguistic experience.

3pSC5. Effects of L2 experience on L2 vowel production and L1 speakers' perception of the improved vowels. Grace E. Oh (English Lit. and Lang., Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 05029, South Korea, gracey1980@yahoo.com)

Effects of L2 experience on the production of L2 vowels were investigated and the perceptual accuracy of the L2 vowels by native speakers was tested to examine whether an improvement in vowel articulation leads to higher accuracy. A total of twenty Mandarin Chinese differing in the experience (6 months vs. 2 years) were compared to ten native Korean speakers in their production of seven Korean vowels, /i,ɛ,ɨ,ʌ,o,u,a/. More experienced Chinese speakers were expected to have acquired new Korean vowels, /ɛ,ɨ,ʌ,o/, in a more native-like manner than the inexperienced group. The results showed that Chinese learners were able to produce the three similar vowels, /i, u, a/ in a native-like manner even before any exposure to Korean. After 2 years of experience, they have shown to produce the new mid vowels, /ɛ, ʌ/ with greater height distinctions (F1 values) from adjacent high vowels. Although the two high back vowels, /o, u/ were deviant from the native norm, the Experienced group learned to distinguish the vowels with a distinctive F2 frequency, which has become a critical cue for the distinction of the two merging vowels by young Seoul speakers. When native Korean speakers were asked to identify the vowels produced by Chinese speakers, not only new mid vowels but also similar vowels were more accurately identified for the experienced group. The greatest improvement in perceptual accuracy was shown for the categories that were newly established by the inexperienced learners.

3pSC6. Foreign accent in L2 Japanese: A cross-sectional study. Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, Eugene, OR 97403, idemaru@uoregon.edu), Kimiko Tsukada (Univ. of Oregon, Sydney, New South Wales, Australia), and Misaki Kato (Univ. of Oregon, Eugene, OR)

The current study examines acoustic sources of foreign accent in second language Japanese produced by American learners across different instructional levels and learning backgrounds. Our prior work has demonstrated that pitch accent, vowel duration, and spectral information of the vowel [e] influence perceived foreign accent in Japanese short sentences produced by intermediate learners, with pitch accent exerting the strongest influence. Building on this prior finding, the current study examines Japanese produced by American learners at the beginning level (n = 10), at the intermediate level (n = 16), American learners who have had early exposure to Japanese (n = 10), native Japanese speakers (n = 10), and ratings of the speech samples by native Japanese listeners (n = 10) to investigate the difference in the extent of perceived accent in their speech and the acoustic sources that influence perceived accent. The results of the current study shed light on issues related to development of second language speech, and the perceptual relevance of the development to lay listeners.

3pSC7. The role of voice familiarity in bilingual speech processing. Monika Molnar (Dept. of Speech-Lang. Pathol., Univ. of Toronto, 500 University Ave., Toronto, ON M5G 1V7, Canada, monika.molnar@utoronto.ca)

Interlocutor context affects proficient bilinguals' spoken language processing. For instance, bilinguals in a visual-auditory lexical decision task are able to predict the context-appropriate language based on the visual cues of interlocutor context (e.g., Molnar *et al.*, 2015). Because it has been also demonstrated that bilinguals, as compared to monolinguals, process talker-voice information more efficiently (Levi, 2017), in the current study we addressed the question whether bilinguals are able to predict context-appropriate language based on voice information alone. First, in a same-different task, English monolingual and bilingual participants were familiarized with the voices of 4 female speakers who either spoke English (shared language across the monolingual and bilingual participants) or Farsi (unknown to both monolingual and bilingual participants). Then, in a lexical decision

task, the participants heard the same 4 voices again, but all of the voices spoke in English this time. We predicted that if the participants established a voice-language link in the first part of the task, then their response times should decrease when they hear an "English-voice" (as opposed to a "Farsi-voice") uttering an English word in the lexical decision task. Accordingly, our preliminary results suggest that the bilinguals' performance is facilitated by the established voice-language link.

3pSC8. Comprehension of accented sentences by young and older listeners. Yu-Jung Lin (Indiana Univ., 800 N Union St. Apt. 405, Bloomington, IN 47408-2230, lin41@indiana.edu)

Previous studies have shown that understanding accented speech requires additional cognitive support. Along this line, the current study aims to test whether the ability to understand accented speech can be influenced by aging and whether older listeners are more sensitive to accented speech. Native speakers of Southern Min from two age groups (18–35, 55 and older) are instructed to listen to sentences produced by two groups of speakers, native and nonnative speakers of Southern Min, answer questions regarding the target words in the sentences produced by these speakers, and then rate speakers' comprehensibility and accentedness. The stimuli for the listening test are produced by speakers during a delayed repetition task, where they repeat the Southern Min sentences after hearing each of them. The target words embedded in these sentences contain phonemes known to be difficult for nonnative speakers (e.g., syllable-initial voiced stops, nasalized vowels). It is hypothesized that older group, compared to younger listeners, will score lower on understanding accented speech, and tend to rate more speech accented and rate less speech comprehensible.

3pSC9. The effect of language experience on lexical tone perception. Xianzhuo Li (Int., College of Chinese Studies, Nanjing Normal Univ., Nanjing, Jiangsu, China) and Zhiyan Gao (English, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, zgao@gmu.edu)

Language experience has often shown to affect speech perception at segmental levels (Holt, 2010). Whether and how language experience affect perception of suprasegmental elements is often subject to dispute (Xu *et al.*, 2006; Peng, 2016). The current study focused on the identification and discrimination of Mandarin lexical tones by 3 groups of listeners, namely, 15 native speakers of Mandarin Chinese (tonal language), 15 native speakers of Lao (tonal language), and 15 native speakers of Uzbek (non-tonal language). We ask (1) whether experience with lexical tones (tonal vs. non-tonal) affects categorical perception of pitch contours; (2) whether differences in native tone inventories (Mandarin vs. Lao) affect pitch perception. Stimuli were 7 monosyllabic snippets resynthesized from a natural Mandarin speech sample. The stimuli represented a physical continuum of pitch contours ranging from a Mandarin level to a Mandarin rising tone. Preliminary results show a strong categorical perception of pitch contours for Mandarin speakers than for Lao and Uzbek speakers, and Lao speakers exhibited stronger categorical perception of pitch contours than Uzbek speakers. The results indicate that lexical tone perception is affected by both the experience with tonal languages and one's native tone inventory.

3pSC10. Perceptual cues used by Chinese native speakers in English /i/-/ɪ/ distinction. Zhuqing Wang (Ewha Womans Univ., Rm. 324, Inwun-guawk, 52 Ewhayeodae-gil, Seodaemun-gu, Seoul, Seoul 03760, South Korea, jadeindurham@ewhain.net)

Native English speakers rely on spectral cues primarily and durational cues secondarily for /i/-/ɪ/ distinction whereas Chinese EFL learners tend to ignore tense-lax markedness due to a lack of lax /ɪ/ in Chinese vowel inventory and the speakers' insensitivity to formant values. Therefore, this study aims to explore whether Chinese speakers can use both durational and spectral cues for /i/-/ɪ/ distinction and whether females and males perform similarly in a perception test. The test was administered to 12 intermediate-level Chinese native speakers. *Beat* and *bit* were used as the basic stimuli and each stimulus was manipulated with seven different durations. The result showed that participants were predisposed to regard stimulus with long durations as /i/ and short durations as /ɪ/. Both the factor of gender and vowel duration had a significant effect on the correctness. Females were outperformed by males in the perception test correctness in most duration points except for very short vowels. Additionally, the size of differences between genders was larger in /ɪ/ than /i/ responses. In conclusion, although no significant spectral cue use was found with the participants, some uses of spectral cues at different vowel durations (90 ms, 390 ms, and 450 ms) were observed.

3pSC11. Exploring the skill of being good with accents: A study of monolingual English speakers' ability to reproduce a novel foreign accent. Laura Spinu (Dept. of Communications and Performing Arts, CUNY Kingsborough Community College, Brooklyn, NY) and Nadya Pincus (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 812 Lehigh Rd., Newark, DE 19711, npincus@udel.edu)

We collected production data from monolingual English speakers who were trained to produce a foreign accent of English, specifically Russian English (RE). First, the subjects read a paragraph in their own accent (baseline). Second, they listened to recordings of English sentences produced by a native speaker of Russian; next, the sentences were replayed, and the subjects imitated each sentence after hearing it. Finally, they were instructed to read the baseline sentences again (without audio prompts), trying their best to reproduce the RE accent. Several acoustic analyses are underway, addressing (a) vowel characteristics, (b) realization of stops, and (c) intonation patterns. Preliminary results show considerable differences in speakers' performance. The questions we address based on the data collected are (1) is there a continuous or more categorical change in speakers' ability to reproduce the RE accent? (2) which aspects of RE are most salient to native English speakers?, and (3) is there a correlation in quality and quantity such that speakers who pick up on more cues also produce them in more native-like manner? Our goal is to develop an algorithm to predict a given speaker's accentedness, and compare its performance with native speakers' accentedness ratings in a follow-up perceptual experiment.

3pSC12. Adult learners' use of lexical cues in the acquisition of L2 allophones. Shannon L. Barrios and Joselyn Rodriguez (Linguist, Univ. of Utah, 255 S.Central Campus Dr., Rm. 2313, Salt Lake City, UT 84108, s.barrios@utah.edu)

Adult second language (L2) learners gain knowledge of L2 allophones with experience. However, it is not well understood *how* this knowledge is acquired. We investigated whether naïve subjects use lexical cues in the form of visual referents to acquire L2 allophones. We exposed native English speakers to one of two artificial languages where novel words containing two acoustically similar sounds, [b] and [β], occurred in an overlapping distribution. The words were paired with an image that either did or did not reinforce the contrast (i.e., [bati]—"apple" and [βati]—"penguin" (*DiffImage* group) or [bati]—"penguin" and [βati]—"penguin" (*SameImage* group)). Participants completed three tests to determine whether the exposure phase impacted their ability to perceive and lexically encode the [b]-[β] contrast in trained and untrained words. If subjects use lexical cues in the form of visual referents to infer the phonological status of [b] and [β], then participants in the *DiffImage* group, but not the *SameImage* group, should discriminate and lexically encode the distinction. Data from 40 participants suggest that lexical cues may impact participants' lexical encoding, but not perceptual sensitivity, to the [b]-[β] contrast. We discuss our findings in relation to proposed learning mechanisms.

3pSC13. The determining factor of L2 vowel perception: The establishment of L2 categories. Wei Hu (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., No. 241, Weijin Rd., Hexi District, Tianjin 300074, China, jykxhuwei@tjnu.edu.cn), Sha Tao (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China), Lihong Wang (School of Educational Sci., Tianjin Normal Univ., Tianjin, China), and Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

The L2 speech learning theories argue that the perception of L2 phonemes is mainly influenced by the relationship between and among L1 and L2 phonemes. In the present study, we hypothesized that L2 vowel identification is primarily determined by the solidity of the L2 vowel category establishment and the perceptual distance between L2 vowel categories. A group of 31 native Mandarin Chinese-speaking listeners and a control group of 9 native English-speaking listeners participated in the study. All listeners were asked to identify natural English/Chinese vowel identification and categorize and rate synthetic English/Chinese vowels. Major findings include: 1) Chinese learners' English vowel perception was associated with their capacity to establish English vowel category; 2) the perceptual distance between English vowel contrasts played a key role in Chinese learners' English vowel perception, rather than the distance between the Chinese vowel and English counterpart. Overall, results of this study suggest that the individual variability of Chinese-native listeners' English vowel perception is interpreted by the establishment of English vowel categories. [Work supported by China Natural Science Foundation 31628009.]

3pSC14. Production and perception of cross-linguistic categories by Polish-English heritage bilingual children. Maximilian Topps (Lancaster Univ., 6 Orchard Pl., Arundel BN18 9BP, United Kingdom, m.topps1@lancaster.ac.uk)

The acquisition of phonetic contrasts relies on acoustic cues such as duration and formant frequencies, which are used to distinguish acoustic signals. For second language (L2) learners, acquiring native-like L2 contrasts can be challenging when they are similar to first language contrasts, or are distinguished by different acoustic cues. Models such as the L2 Language Perception model (L2LP; Escudero, 2005), Perceptual Assimilation Model (PAM-L2; Best & Tyler, 2007), and the Speech Learning Model (SLM; Flege, 1995) make predictions regarding the outcomes of cross-linguistically similar contrast acquisition based on phonetic perception. This study investigates the similar contrasts of English /i/-/ɪ/ and Polish /i/-/ɨ/ acquired by English-Polish heritage bilingual children. The objective is to observe the models' predictions in the context of young bilinguals, and to compare strategies across perception and production. Data were collected from 18 children aged 5;11 to 10;7. For each contrast, a picture-naming task elicited production data, and a categorisation task, involving a phonetic continuum across a lexical minimal pair, elicited perception data. Results suggest that unique strategies are used in production for each language. However, this was not reflected in perception data, which suggests perceptual assimilation. Results are discussed within a framework of relative acoustic cue weighting.

3pSC15. A study on the vowel duration difference followed by phonetic contexts. Sooyoung Lee (English Lang. and Lit., Yonsei Univ., 207, 367-29, Seongsan-ro, Seoul, Seodaemun-gu 03726, South Korea, sree0910@naver.com) and Seok-Chae Lee (English Lang. and Lit., Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

Vowel duration in English serves as a cue to the voicing distinction of the following consonant, in a way that vowels preceding voiced consonants are longer than those preceding voiceless consonants. This study will examine the differences in vowel duration in various phonetic contexts produced by 8 Korean-speaking L2 learners of English in comparison with the differences produced by 4 native English speakers from the K-SEC corpus (Korean-Spoken English corpus). Specifically, we will test for the durational evidence of the influence of different manners of articulation and syllable positions on preceding or following vowels and investigate how the influence is phonetically realized by two language groups. Our hypotheses and predictions are as follows: 1) vowels are longer when preceded by consonants in coda position than when followed by the same consonants in onset position, 2) vowels are longer when preceded by nasals than when preceded by non-nasal consonants, and 3) Korean speakers produce the differences in vowel length to a smaller extent than native English speakers.

3pSC16. A multigenerational investigation of the acoustics of English-Modern Hebrew heritage speakers. Kyle Jones (Univ. of Arizona, 3015 Brunnet Ln., Sacramento, CA 95833, kysjones@email.arizona.edu)

This research investigates the speech acoustics of two generations of U.S. *olim* (immigrants) in Israel: First generation immigrants, whose first language (L1) is American English (AE), and their second-generation children, for whom English is a heritage language (HL). A specific HL accent has been demonstrated in the studies that have investigated the phenomenon, showing that heritage speakers have good control of phonetic/phonological contrasts between their two languages and distinct patterns from both NS and L2 learners (who both show L1 influence). The research focuses on issues of heritage language phonology and intergenerational multilingualism: What is the speech of HL speakers of AE in Israel like? How does this speech compare to the speech of their parents (their main source of input for AE)? How does Modern Hebrew (MH), their L2 or primary language, affect their AE? These questions are investigated through a language questionnaire and an acoustic analysis of voice onset time (VOT) in stops. Acoustic analysis demonstrates that HL speakers, echoing previous studies, have excellent control over phonetic and phonological contrasts in salient distinctions between their two languages, despite greater overall variability. VOT is closer to MH norms when speaking MH and closer to AE norms when speaking AE.

3pSC17. Perception of spectrally-shifted non-native speech. Michelle R. Kapolowicz (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, 800 West Campbell GR 4.1, Richardson, TX 75080, michelle.kapolowicz@utdallas.edu), Daniel Guest (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), Vahid Montazeri (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, Richardson, TX), Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR), and Peter F. Assmann (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, Richardson, TX)

Some studies have suggested that perceiving speech from multiple non-native talkers can be more difficult than perceiving speech from a single non-native talker. We found evidence in favor of this idea, previously showing a detrimental effect on sentence intelligibility of reducing the spectral resolution of non-native speech via a 9-channel tone vocoder (a condition where talker-specific spectral cues are limited). This effect was more severe when perceiving non-native speech from multiple talkers compared to a single talker, whereas we observed no differences across single- and multiple-native talker conditions. These results imply that listeners place greater reliance on talker-specific spectral cues to aid with perception of non-native speech as compared to native speech. In the present study, we further examined the effects of talker-specific spectral cues on intelligibility by scaling the spectral envelope and fundamental frequency (F0) of previously recorded sentences in successive increments of 8% and 30%, respectively, using the STRAIGHT vocoder. As predicted, intelligibility of spectrally-shifted sentences spoken from a single, non-native talker was similar to that observed for multiple non-native talkers, and listeners reported hearing spectrally-shifted speech as coming from different talkers. These results confirm the importance of spectral cues for the perception of non-native speech.

3pSC18. Lexical representation of Mandarin tones in second language learners. Kuo-Chan Sun (Univ. of AB, 3-20 Pembina Hall, University of AB, Edmonton, AB T6G 2H8, Canada, kuochan@ualberta.ca)

Previous studies have shown that a word's phonological similarity to other words (i.e., phonological neighborhood) can influence its recognition. However, most research concerning lexical representations has been observed for neighbors based on segmental overlap, and little is known about such effects with suprasegmentals such as Mandarin tones. In the present study, two experiments were conducted with forty L2 listeners and 40 native speakers to examine how tone neighborhood density influences Mandarin spoken word recognition. In Experiment 1, speed and accuracy

from both groups' performance in an auditory lexical task were influenced by tone neighborhood density (i.e., fewer words were recognized from dense tone neighborhoods than from sparse tone neighborhoods). However, L2 listeners' performance was inferior to native listeners'. In Experiment 2, form priming patterns showed that reliable facilitation was observed only when the prime and the target were identical, while monosyllabic Mandarin words differing only in tone failed to speed the response to the target. In addition, only L2 listeners showed an increase in RTs to respond to the target when it was preceded by tone overlap primes. The results of these experiments demonstrate that tone neighborhood is an important factor in L2 Mandarin spoken word recognition.

3pSC19. Effects of audiovisual perceptual training with corrective feedback on the perception and production of English sounds by Korean learners. Ho-Young Lee (Linguist, Seoul National Univ., Seoul, South Korea), Jookyeong Lee (English Lang. and Lit., Univ. of Seoul, Seoul, South Korea), Hyosung Hwang, and Joo Yeun Lim (Linguist, Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, hshwang415@gmail.com)

This paper aims to propose a new audiovisual perceptual training with corrective feedback and compare the results of this training with those of audio only and audiovisual perceptual training. The results are compared in order to see which is more effective in improving the perception and production of English sounds by Korean learners. For this experiment, we selected 6 consonant pairs (i.e., /p-f/, /b-v/, /t-θ/, /l-r/, /s-ʃ/) and 4 vowel pairs (i.e., /iː-ɪ/, /ɛ-æ/, /ɑː-ʌ/, /ɔː-oʊ/), which are visually distinguishable from one another but often confused by Koreans. 20 subjects participated in audio only training (A group), 20 subjects in audiovisual training (AV group) and 25 subjects in audiovisual training with corrective feedback (CF group). 8 training sessions, each lasting about 40 minutes, were offered. All the groups showed a significant improvement in the perception of vowels and consonants (i.e., 5.4% and 3.4% improvements for the A group, 7.7% and 3.4% for the AV group, and 11.1% and 4.1% for the CF group). However, only A group showed a significant improvement in the production of vowels (i.e., 7.19% improvement) while only AV and CF groups showed a significant improvement in the production of consonants (i.e. 6.08% and 8.00% improvements for the AV and CF groups, respectively).

3pSC20. Listener flexibility to lexical alterations for foreign- and native-accented speech. Kali Burke (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Michelle K. Tulloch (Psych., Florida Atlantic Univ., Buffalo, NY), and Marieke van Heugten (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Listeners regularly encounter speech produced by speakers born and raised in linguistic communities different from their own. Given that individuals from different linguistic communities often speak in a distinct fashion, it is not uncommon for listeners to be exposed to accent-induced variability. This can lead to discrepancies in the realization of words, rendering such speakers difficult to understand. As a result, listeners tend to increase their flexibility in their interpretation of variations in the phonological form for known words. To examine whether or not a foreign accent also leads to an increased flexibility in label-object mapping, we tested participants in a preferential looking paradigm. Using a between-subject design, participants either listened to a foreign- or to a native-accented speaker. In some trials, the target word labeled an object that was depicted on the screen (match trials), while in other trials the target word referred to an object that was closely related (but not an exact match) to one of the objects depicted on the screen (mismatch trials). Listeners' looking behavior after target word onset was examined across accents and trial types. The results of this study show that lexical alterations affect word recognition across foreign- and native-accented speech.

3pSC21. Production of English stop voicing distinction in familiar and unfamiliar positions by Spanish speakers. Alicia Swain and Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover Ctr. W252, Athens, OH 45701, as569911@ohio.edu)

The voicing distinction in English stop consonants involves distinct acoustic correlates depending on the position of the consonant. For Spanish speakers, prevocalic voicing distinction is in a familiar position whereas postvocalic voicing is in an unfamiliar position. In this study we examined whether Spanish speakers could use voice onset time and vowel duration to produce the voicing distinction in these two positions. Acoustic analysis showed Spanish speakers were able to produce the voicing distinction in both positions, although the magnitude of between-category difference is smaller compared to those produced by native English speakers. Production of the VOT distinction for prevocalic stops improved significantly following a brief tutorial on the acoustics of stop consonants, but the vowel duration difference for postvocalic stops did not show comparable change. These results suggest implementation of a phonological contrast in an unfamiliar position is constrained by native language characteristics. [Work supported by Ohio University College of Health Sciences and Professions Student Research Grant.]

3pSC22. Emotional responses to speech accommodation: A systematic review. Kathrin Rothermich, Havan Harris (Commun. Sci. and Disord., East Carolina Univ., 355 Beasley Dr., F3, Greenville, NC 27834, rothermichk17@ecu.edu), Kerry Sewell (Laupus Health Sci. Library, East Carolina Univ., Greenville, NC), and Susan Bobb (Psych., Gordon College, Wenham, MA)

Five percent of the U.S. population speak English "not well" or not at all, often leading to miscommunication, which can be especially problematic in a healthcare environment or at the workplace. According to socio-linguistic frameworks such as Communication Accommodation Theory, native speakers have interactional goals and strategies to communicate effectively with non-native speakers. One of these strategies, foreign-directed speech, involves modifying the speech output acoustically and linguistically by slowing down, simplifying speech, and exaggerating vowels. While the acoustic properties of foreign-directed speech are well documented, research is limited on how non-native speakers interpret them emotionally and pragmatically. For instance, do non-native speakers find foreign-direct speech helpful or does it come across as condescending? The purpose of this systematic review is to determine the current evidence on how speech accommodation affects the listener, with a special focus on how it is perceived by non-native speakers. In this review, we outline the basic components of communication accommodation, provide a systematic review of the literature related to the emotional effects of speech accommodation, and discuss current issues. We conclude by formulating recommendations for future research.

3pSC23. Acquisition of categorical perception of Mandarin tone. Michelle X. Li (Office of Linguist, Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, xiao.xiao.li@mail.mcgill.ca), Daniel Rivas (Psych., Université du Québec à Montréal, Montreal, QC, Canada), Fernanda Pérez Gay Juárez (Neurosci., McGill Univ., Montreal, QC, Canada), Tomy Sicotte, and Stevan Harnad (Psych., Université du Québec à Montréal, Montreal, QC, Canada)

This experiment examined the effect of motor repetition on the learning of Mandarin pitch categories by non-native speakers. In Mandarin, there are four tonal categories, which refer to pitch contours that discriminate words. Differentiating tonal categories is both essential and arduous for non-native speakers to learn. Native speakers do this effortlessly because they have a categorical perception (CP) effect for tones, i.e. they perceive items within a tonal category as more similar to each other and items between categories as more different. Non-native speakers do not have this effect, and this experiment attempted to induce the CP effect for Mandarin tones via a training task, which was the experimental manipulation: Participants either repeated a tone stimulus before categorizing it or listened to a stimulus and categorized it without repetition. Discrimination between and within tonal categories was measured before and after training. All participants demonstrated increased between-category and within-category discrimination after training, except for learners who repeated stimuli in the training phase. They demonstrated a decrease in within-category discrimination, showing a weak CP effect that could be stronger with more training. Implications of these results on auditory category learning and language education will be discussed.

3pSC24. Effects of the place of articulation of the following consonant on the identification and discrimination of American English vowels by native speakers of Japanese and Korean. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Heesun Han (Osaka Univ., Toyonaka, Osaka, Japan)

Native speakers of Japanese and Korean identified 6 American English monophthongs /i, ɪ, ɛ, æ, ɑ, ʌ/ and discriminated 6 vowel pairs /i/-/ɪ/, /ɛ/-/ɪ/, /æ/-/ɛ/, /æ/-/ɑ/, /æ/-/ʌ/, /ɑ/-/ʌ/ in in /hVt/, /pVt/, /pVn/ and /pVl/ frames. The two groups of listeners' identification and discrimination accuracy were submitted to 2 mixed-design ANOVAs, respectively, with 2 Listener Groups as a between-subject variable, and 6 frames and 6 vowels (or vowel pairs) as within-subject variables. As for identification, a main effect of vowel and frame are both significant ($p<.001$), and a three-way interaction of listener groups $\times$ vowels $\times$ frames is also significant ($p<.001$). Post-hoc pair-wise comparisons revealed that Korean listeners identified /ʌ/ significantly better than Japanese listeners. This is probably because Korean listeners can equate the English vowel to Korean /ʌ/. Japanese listeners' identification accuracy of /i/ is lower before /n/ and /l/, but that of Korean listeners was not affected by the context. As for discrimination, a main effect of the frames and vowel pairs was significant ($p<.001$). A three-way interaction of the 3 factors was not significant, but all the two-way interactions were significant ($p<.001$). Post-hoc pair-wise comparisons revealed that Japanese listeners' discrimination accuracy of /i/-/ɪ/ and /æ/-/ɛ/ is lower before /n/ and /l/ than before /t/, while that of Korean listeners was not affected by the context. Korean listeners discriminated /æ/-/ɛ/ least accurately of all the six vowel pairs.

3pSC25. Information-theoretic variables in Spanish-English bilingual speech. Khia A. Johnson (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, khia.johnson@alumni.ubc.ca)

Languages lenite similar segments to different extents—a finding that can be accounted for with information-theoretic variables like frequency, predictability, and informativity [Cohen Priva, 2017, Language 93: 569–597]. Prior research addresses the role of segment information content across languages, but assumes that information-theoretic variables operate on an in-language basis. While appropriate for monolingual speech, this assumption is problematic for bilingual speech, as an individual's languages are known to influence one another (e.g. [Fricke *et al.*, 2016, J. Mem. Lang. 89: 110–137]). In this paper, I report on a study using the Bangor Miami corpus [Deuchar *et al.*, 2014, Advances in the Study of Bilingualism: 93–110], addressing how well information-theoretic variables predict lenition in Spanish-English bilingual speech. Specifically, do they operate on an in-language or cross-language basis? To address this, I focus on the duration of word-medial intervocalic fricatives shared by both languages—/f/ and /s/. Accounting for variables known to affect segment duration (e.g. speech rate), I use linear mixed-effect models to assess the contribution of information-theoretic variables in accounting for duration variation in bilingual speech. Four models are evaluated, compared, and discussed: (i) English in-language, (ii) English cross-language, (iii) Spanish in-language, and (iv) Spanish cross-language.

3pSC26. The effect of native language on the second language vowel variability. Jaekoo Kang (Speech-Language-Hearing Sci. program, CUNY Graduate Ctr., 3547 34th St., Apt. 1E, Long Island City, NY 11106, jkang@gradcenter.cuny.edu), D. H. Whalen (Speech-Language-Hearing Sci. program, CUNY Graduate Ctr., New York, NY), and Hosung Nam (English Lang. and Lit., Korea Univ., Seoul, South Korea)

Korean learners of English must create four vowel categories for English (/i, ɪ/ and /ɛ, æ/) in relation to two similar native categories (/i/ and /ɛ/). It is hypothesized that new categories should be easier to learn than similar ones (Flege, 1994), but it is unclear whether the English L2 vowels are similar or new. The degree of similarity between the four English vowels and the two Korean vowels was examined using the distribution metrics (i.e., ellipse overlap, cross-entropy, and Gaussian Mixture Model) as well as Euclidean distance in F1/F2 space. The L2 spoken corpus included 100 repetitions of words in both Korean and English spoken by 37 Korean L2 learners (20 female). Preliminary results indicate that the English (L2) high vowel pair was more overlapped with Korean /i/ than the low vowel pair with Korean /ɛ/, especially for male speakers. For the English (L2) low vowel pair, female speakers showed less overlap but higher variability along F1 direction than male speakers. This demonstrates that the similarity between Korean and English vowels is characterized by the distribution as well as the distances between the vowel categories. Acoustic results will be further compared with identification by native English speakers.

3pSC27. Difference of articulatory movement between native and non-native consonant clusters. Seiya Funatsu (Sci. Information Ctr., Prefectural Univ. of Hiroshima, 1-1-71 Ujinahigashi Minami-ku, Hiroshima 734-8558, Japan, funatsu@pu-hiroshima.ac.jp) and Masako Fujimoto (Waseda Univ., Tokorozawa, Saitama, Japan)

We investigated the difference of articulatory movement between native and non-native consonant clusters. English has consonant clusters, but Japanese does not. Therefore, speakers chosen for comparison were native English and native Japanese speakers. Speech samples consisted of 4 words, "blat," "bnat," "plat," "pnat." In these words, /bl/ and /pl/ are English clusters, but /bn/ and /pn/ are not. We measured the movement of the tongue tip, the mandible and the lower lip by WAVE system (NDI corp.). There were remarkable differences in the mandible and the lower lip movement between native (/bl/, /pl/) and non-native (/bn/, /pn/) clusters in English speakers. Namely, with the non-native clusters the difference of the articulatory movement in the mandible and the lower lip of every utterance was quite large; however, in native clusters, the difference was quite small. For Japanese speakers it was large for all clusters. Thus, it was revealed that the articulatory movement of the mandible and the lower lip in non-native clusters was not stable in native English speakers, even though English has consonant clusters. (This study was supported by KAKENHI 15K02524, 17K02707.)

3pSC28. An ultrasound investigation of how accurately people follow tongue movement instructions. Ian Wilson (CLR Phonet. Lab, Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, wilson@u-aizu.ac.jp) and Julián Villegas (Comput. Arts Lab., Univ. of Aizu, Aizu Wakamtsu, Fukushima, Japan)

In an analytic-linguistic approach to teaching segmental pronunciation and articulatory setting, teachers and voice coaches give explicit instructions on tongue placement and movements. Instructors assume that learners can do exactly as instructed. This assumption was tested in research by Wilson and Horiguchi (2012, PSLLT), who showed that phonetically untrained participants were very poor at following explicit tongue movement instructions. In their study, both the magnitude and direction of movement of the tongue's centre of gravity were calculated from 2D ultrasound images. However, by only measuring changes in the centre of gravity, it is possible that movements were found to be smaller than they really were, especially if participants focused on the front of the tongue rather than the whole tongue

body. In this study, we reanalyzed the original data, this time focusing on the surface of the tongue, rather than the centre of gravity. We made tongue traces using EdgeTrak software (Li *et al*., 2005), and compared them using a Smoothing-Spline ANOVA method (Davidson, 2006). Results differed from the original study, showing that participants were more conscious of what the front of the tongue was doing rather than the whole tongue body. Implications for segmental pronunciation teaching will be discussed.

3pSC29. The effectiveness of audio-visual training on non-native English speech production and perception. Chia-Ni Shen (Linguist, Univ. of Oregon, 17460 SW 104th Ave., Tualatin, OR 97062, jshen@uoregon.edu) and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR)

This project examines the effectiveness of audio-visual training on non-native English speech production and perception. Previous research utilizing audio-visual training has been employed in the field of speech pathology, showing positive outcomes in improving speech among dyslexic children. However, few studies to date have examined its use in second language learning, specifically bilabial and labiodental consonants (i.e., /b, p, m, f, v/), which are known to be challenging for many second language learners. The aim of this project is to explore audio-visual training across 3 native language groups, Mandarin Chinese, Japanese, and Arabic, who are all English language learners. Participants undergo a training regimen designed to examine the effects of audio-visual and audio-only training. Performance before and after training is assessed via perception and production tests. We hypothesize that 1) production and perception performance will improve after two days of training and will improve more after audio-visual training and 2) production and perception improvement will rely heavily on a participants' language background and known difficulties with specific bilabial and labiodental sounds. Results from this study will enrich understanding of the effectiveness of audio-visual training in second language learning.

3pSC30. Parallel adjustment of phonetic targets in L2 English voice onset time. Eleanor Chodroff (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, eleanor.chodroff@northwestern.edu) and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR)

The phonetic realization of any given speech sound varies considerably across speakers and languages. For instance, the voice onset time (VOT) of [k^h] can range from ~40 ms to over 100 ms (Cho & Ladefoged, 1999). Within American English, recent research has demonstrated that the VOT of [k^h] is also systematically and linearly related to the VOTs of [p^h] and [t^h] (Chodroff & Wilson, 2017). In the present study, we investigated whether these relations were maintained across L2 speakers of English. L2 speech production could arise from independent acquisition (or adjustment) of the phonetic targets for each speech segment. Alternatively, the presence of covariation would indicate that properties of speech sounds may be altered in tandem. To investigate this, VOT data was obtained from a subset of the ALLSSTAR Corpus, which contained matched connected speech data from 26 L1 American English speakers and 114 L2 English speakers (22 unique L1s). Preliminary analysis revealed strong VOT covariation among aspirated stops across L2 speakers (rs = 0.67 to 0.74), and qualitatively equivalent covariation across L1 speakers (rs = 0.57 to 0.75). The observed covariation may arise from a universal principle of uniformity requiring near-identical implementation of the shared laryngeal feature value.

3pSC31. A study of the difference vowel duration according to pronunciation assessment and the stress. Kihoon Park (Yonsei Univ., Yonsei University Graduated School of Education, Seoul, Seodaemun-gu 120-749, South Korea, manim0317@naver.com) and Seok-Chae Lee (Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

The purpose of this paper is to examine Korean elementary English learners' production of lexical and phrasal stress and the difference of the vowel length according to stress and pronunciation assessment. For the study, Korean-spoken English Corpus spoken by Korean elementary students was used. Lexical and phrasal stress were analyzed in terms of vowel length. As a result of the analysis, the following was revealed. A) Higher level groups of pronunciation assessment pronounced stressed syllables much longer than unstressed syllables compared to the lower level groups. B) As a result of the comparison according to the stress position, it was found that the stressed vowel located in the second syllable was pronounced much longer in all the pronunciation assessment group. It can be predicted as a result of the effect of mother tongue interference, which does not allow diphthong. C) Speakers that produced stressed syllables with much longer duration than unstressed syllables were classified as higher level groups in assessment. The result of this research suggests that segments and supra segments should be taught at the same time.

3pSC32. Quantifying fine-scale details of vowel spaces in German language learners. Lauren Elliott (Psych., Carthage College, 2001 Alford Park Dr., Kenosha, WI 53140, lelliott1@carthage.edu) and Benjamin N. Taft (Landmark Acoust. LLC, Racine, WI)

Second language learners may be particularly challenged when the new language makes distinctions between sounds that are phonetically equivalent in the speaker's first language. If the distinction occurs between vowels, native speakers and learners should occupy the relevant vowel spaces in different ways. Native speakers' vowel spaces should show denser, more acoustically separate regions for each vowel. Learners' spaces should show less consistent, more overlapping spaces. In contrast, when a pair of vowels are phonetically distinct in both languages, there should be little difference between native and learner acoustic spaces. We use a custom formant tracker to quantify the vowel spaces of two native speakers and 6 German language learners.

3pSC33. Digit span error patterns in bilinguals and monolinguals. Noah M. Philipp-Muller (Linguist, Univ. of Toronto, 4th Fl., 100 St. George St., Toronto, ON M5S3G3, Canada, noah.philippmuller@gmail.com), Laura Spinu (Communications and Performing Arts, CUNY, New York, NY), and Yasaman Rafat (Modern Lang. and Literatures, Western Univ., London, ON, Canada)

Research shows that bilinguals tend to outperform monolinguals on certain cognitive and linguistic tasks. While the mechanism underlying these advantages remains unclear, it has been suggested that bilinguals have enhanced working memory, which may be responsible for some of the cognitive advantages observed in these populations. To examine the mechanism responsible for the bilingual advantage, serial working memory was compared between monolingual and bilingual undergraduate students using an adaptive digit span test ($n = 77$). The results of the test were algorithmically adjusted in order to reveal not only if each digit was correct, but also the existence of serial errors and digit scrambling. The results showed that bilinguals made significantly fewer transpositional errors compared to monolinguals ($p < 0.02$, $d = 0.61$). These results are thought to be caused by differences in attention to serial information between bilinguals and monolinguals. Another possible explanation for these results is that bilinguals implement more cognitive memory techniques (like chunking), and are therefore better at retaining serial information. The computational methods developed in this experiment will help guide paths for further research on the impact of syntactical co-activation on serial memory recall, and primacy/recency effects.

Session 3pSPa

Signal Processing in Acoustics: General Topics in Signal Processing I (Poster Session)

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

All posters will be on display and all author will be at their posters from 1:00 p.m. to 3:30 p.m.

Contributed Papers

3pSPa1. Assessing multimodal speech-gesture coordination: correlation map analysis for signal comparison. Samantha G. Danner (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, sfgordon@usc.edu)

Speech acoustics and body movements are systematically linked during speaking; however, there is no consensus regarding which acoustic and/or kinematic signals are relevant for this coordination nor what critical landmarks facilitate this linkage. To address this, we use correlation map analysis (CMA) of speech acoustics and body movement to assess the hypothesis that speech and gesture are coordinated at prosodically prominent speech regions. CMA allows for the analysis of correlation between any two continuous signals both instantaneously and at user-defined ranges of delay. In this study, we examine the relationship of the speech's RMS acoustic amplitude signal with the dominant hand's velocity signal at speech turns, phrase boundaries, acoustic amplitude peaks, and other ("elsewhere") regions. We find that turns and phrase edges exhibit the greatest likelihood of positive correlation. Additionally, the likelihood of correlation is higher when the hand velocity signal is delayed with respect to the amplitude signal. These results suggest that speech and gesture may be strongly linked at speech turn landmarks for the purpose of signaling a floor exchange. Thus in addition to coordination for semantic and prosodic purposes, we propose that speech-gesture coordination can serve with prosody to signal speech turns in conversation. [Work supported by NIH.]

3pSPa2. Compressed sensing and recovery of underwater acoustic signal in Internet of Things. Feiyun Wu, Kunde Yang, Quan Sun, and Yunchao Zhu (Northwestern PolyTech. Univ., Youyi Rd 127, Xi'an, Shaanxi 710072, China, wfy@nwpu.edu.cn)

Telemonitoring and communications technologies of underwater information is inseparable from the process of sampling and transmitting of data. Conventional methods often fail in energy efficiency to obtain such a huge data in Internet of Things (IoT), compressed sensing (CS) provides a new perspective to solve the problem. Unfortunately, the underwater acoustic signal is non-sparse in the time domain and thus the current CS methods cannot be used directly. This study adopts the fast Fourier transform (FFT) based dictionary-matrix for sparse representation. Then we design an approach based on an approximated l_0 (AL0) norm at the receiving terminal, to search the sparse solution via the filtered steepest descent method and projections. The sparse estimation is used for reconstruction of collected data, combing with the previously used measurement matrix and dictionary-matrix. Experimental results confirm the superior performances of the strategies of the proposed method than the traditional methods including matching pursuit (MP) and orthogonal matching pursuit (OMP) methods.

3pSPa3. Experimental study of efficient multi-band underwater acoustic communication algorithm. Hui Su Lee, Chang Uk Baek, Jung-Hyun Seo, Ji Won Jung (Radio Commun. and Eng., Korea Maritime and Ocean Univ., 430,College of Engineering1, 727, Taejong-ro, Yeongdo-gu, Busan 49112, South Korea, lhs6778@kmou.ac.kr), and Dae Won Do (Agency for Defense Development, Changwon, South Korea)

In the underwater acoustic communication environment, multipath transfer characteristics or Doppler spread due to time and space changes such as seabed, sea level, and water depth affect the performance. The multi-band communication technique is effective in terms of performance and throughput efficiency because it can overcome selective frequency fading by allocating the same data to different frequency bands in the environment of rapidly changing channel transfer characteristic. In addition, the transmission distance can be further extended while overcoming various underwater channel environments. However, the multi-band configuration may have worse performance than the single-band one. It is because the performance degradation in a particular band affects the output from the entire bands, which is input into a decoder, thereby decreasing the overall performance. This problem can be solved through a receiving end that analyzes error rates of each band, sets threshold values, and allocates lower weights to inferior bands. In this paper, we proposed an algorithm to set the threshold value using the preamble error rate, which is known data to be transmitted and received. We have analyzed the efficiency of multi-band transmission scheme by applying 1–4 number of multi-bands using turbo pi code through actual underwater experiment.

3pSPa4. Study on the structure of an efficient receiver for multi-sensors using direct sequence spread spectrum. Ahyun Lee, Chang Uk Baek, Ji Won Jung (Radio Commun. and Eng., Korea Maritime and Ocean Univ., Yeongdo-gu, Busan, Busan 13557, South Korea, ahean@kmou.ac.kr), and Dae Won Do (Agency for Defense Development, Changwon, South Korea)

In underwater acoustic communication, there has been a lot of interest in detection and acquisition of information about multiple sensors as well as detection of information on a single sensor. However, research on multiple sensors access is lacking. In order to acquire information from each sensor in multiple sensors access underwater communication, we mainly use the spreading method which can obtain information of each sensor individually at the receiving side by multiplying the code with different orthogonal components in the same frequency band. In the case of multiple sensors, successive interference cancellation is used to remove interference from other sensor and apply the RAKE method. However, in a multiple sensor communication environment, if the channel information is not perfect, the performance is reduced. In this paper, we propose a transceiver structure for

multiple sensor covert acoustic communication using the spread spectrum method and RAKE method, and propose and effective receiver structure to overcome drawbacks. An actual underwater experiment was conducted to analyze the performance of the covert underwater acoustic communication. The result of experiment indicated that the transmission and receive structure proposed through the actual underwater experiment was well-suited to the communication model for covert multiple accesses.

3pSPa5. Ultrasonic through-tissue communication with video capabilities. Gizem Tabak, Michael Oelze (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 119 Coordinated Sci. Lab., 1308 W Main St., Urbana, IL 61801, tabak2@illinois.edu), and Andrew Singer (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The use of wireless implanted medical devices (IMDs), which communicate data wirelessly from sensors within the body to a receiver outside of the body, are gaining significant momentum in medical diagnosis and treatment procedures. Currently, radio frequency (RF) electromagnetic waves are the most frequently used communication method for wireless IMDs. However, there are various drawbacks of using RF electromagnetic waves in such applications such as high attenuation in the body and strictly regulated frequency spectrum, which consequently limit data rates of these devices. An alternative approach to RF transmission for IMDs uses ultrasonic waves, which experience lower attenuation in the body and have higher available bandwidth. In our work, we aim to demonstrate high data rate ($>$1.2 Mbps) through tissue communication using ultrasonic waves that will allow us to stream video with an IMD while actively controlling it inside the body. Initial experiments performed with 2mm biocompatible sonomicrometry transducers, where 2.4 Mbps data rate BER less than 5E-5 through water is achieved, demonstrating the capability of this method for high speed data transmission. Additional targets of this work include data transmission through beef liver and a live rabbit abdominal wall with high enough data rates capable of video streaming.

3pSPa6. Nonlinear waveform distortion: Assessment and detection of clipping on speech data and systems. John H. L. Hansen, Allen Stauffer, and Wei Xia (Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Speech, speaker, and language systems have traditionally relied on carefully collected speech material for training acoustic models. There is an overwhelming abundance of publicly accessible audio material available for training. A major challenge, however, is that such found data is not professionally recorded, and therefore may contain a wide diversity of background noise, nonlinear distortions, or other unknown environmental or technology-based contamination or mismatch. There is a critical need for automatic analysis to screen such unknown data sets before acoustic model development training, or to perform input audio purity screening prior to classification. In this study, we propose a waveform based clipping detection algorithm for naturalistic audio streams and analyze the impact of clipping at different severities on speech quality measures and automatic speaker identification systems. We use the TIMIT and NIST-SRE08 speech corpora as case studies. The results show, as expected, that clipping introduces a nonlinear distortion into clean speech data, which reduces speech quality and performance for speaker recognition. We also investigate what degree of clipping can be present to sustain effective speech system performance. The proposed detection system, which will be released, could potentially contribute to new audio collections for speech and language technology development.

3pSPa7. Numerical analysis for source direction finding using a cylindrical array with scattered sound fields. Sea-Moon Kim (Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312beon-gil, Yuseong-gu, Daejeon 34103, South Korea, smkim@kriso.re.kr), Keunhwa Lee (Dept. of Defense Systems Eng., Sejong Univ., Seoul, South Korea), and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

Numerous studies on beamforming techniques have been done for source direction finding using various types of arrays. A circular array as well as a linear array is also widely used for its compact configuration and smaller angular dependency of beamforming performance. In underwater environments, a circular array faces high drag forces due to not only ocean current but also its moving platform. One of the solutions may be an array of hydrophones flush mounted on a cylindrical structure to reduce the drag. However this configuration causes sound scattering, which deteriorate the pressure distribution based on the free field condition. This talk deals with numerical approach for source localization techniques with scattered sound fields. Parametric study shows that beamforming performance is improved and relative high resolution can be achieved using an array of small-sized aperture. [This work was financially supported by the research project PES9410 funded by KRISO.]

3pSPa8. High-resolution ultrasound camera with sparse array for machine diagnosis. Choon-Su Park (Ctr. for Safety Measurements, Korea Res. Inst. of Standards and Sci., Gajeong-ro 267, Bldg. # 206 / Rm. # 206, Daejeon 34113, South Korea, choonsu.park@kriss.re.kr)

Fault detection plays a crucial role in maintenance of mechanical systems. Ultrasound is known to be a prevalent feature of incipient fault. Industrial ultrasound detectors are widely used to find out initial malfunction of machinery. Especially, ultrasound camera can shows the fault by overlapping beamforming power distribution with optical camera image. One can easily observe where the faults are. Ultrasound transducer usually has larger size than the wavelength of its resonance frequency for highly sensitive measurement. It must be a useful approach to design sparse array that gets over the spatial aliasing problem. The aliased beamforming distribution contains high grating-lobe level due to rough spatial sampling beyond half wavenumber spacing. The sparse array is optimized to minimize maximum side-lobe level by genetic algorithm under given constraints. Being with the constraints of size and number of elements, a limitation on spatial resolution is inevitable. Functional beamforming is simple but effective to dramatically increase spatial resolution as well as reduce side-lobes. A high resolution imaging scheme based on functional beamforming with an optimal sparse array is introduced.

3pSPa9. Acoustic and ultrasound investigation of word-initial gemination in Moroccan Arabic. Mohamed Yassine Frej (The MARCS Inst., Western Sydney Univ., 2 Bullebourt Ave., Milperra, NSW 2214, Australia, y.frej@westernsydney.edu.au), Christopher Caringnan (Inst. of Phonet. and Speech Processing (Ludwig-Maximilians-Universität München), Munich, Germany), Michael I. Proctor (Dept. of Linguist, ARC Ctr. of Excellence in Cognition and Its Disord., Macquarie Univ., NSW, Australia), and Catherine T Best (The MARCS Inst., Western Sydney Univ., Milperra, NSW, Australia)

Moroccan Arabic uses geminate/singleton contrasts in medial position but it is controversial whether it maintains them utterance-initially. To address this issue, we made simultaneous ultrasound and acoustic recordings of five native speakers producing target words containing /t/-/tt/ and /d/-/dd/ contrasts utterance-initially and -medially, 10 times each. The ultrasound data were analysed via a novel method of using pixel-derived principal components and linear discrimination to generate a time-varying articulatory "closure" signal directly from the ultrasound images, without the need for manual tracing. This articulatory signal was subsequently used to measure closure duration by determining gestural landmarks from its velocity function. The results provide clear *articulatory* evidence that speakers produce utterance-initial singletons versus geminates with significantly different closure durations, although the contrast provides no *acoustic* evidence of closure duration per se. Rather, our results reveal that speakers realize geminate/singleton contrasts via acoustic dimensions not related to consonant duration: The vowel is significantly longer following geminates than singletons, and the stop bursts have greater amplitude for geminates than singletons. Together these findings provide evidence that the gemination contrast is maintained in initial position in Moroccan Arabic, and challenge traditional assumptions that consonant length contrasts are primarily carried by acoustic closure duration differences.

3pSPa10. SoilComm: A miniaturized through-soil wireless data transmission system. Sijung Yang (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1308 West Main St., Coordinated Sci. Lab., Urbana, IL 61801, Syang103@illinois.edu), Omar Baltaji, Youssef M. Hashash (Civil and Environ. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Andrew Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Wireless underground sensor networks (WUSN), are emerging Internet-of-things (IoT) technologies, that can benefit numerous applications including geotechnical data acquisition for online infrastructural health monitoring and automated agricultural systems. State-of-the-art approaches to wireless underground sensor communications have employed, as their data carrier, electromagnetic waves, characterized by: their extreme signal path losses in soil due to absorption and scattering, the challenges they impose on transceiver miniaturization, and the power/energy efficiency constraints they suffer from during battery-based operation. Acoustic waves can resolve many of these challenges, benefiting from lower levels of absorption and scattering losses at target frequencies, and from looser size limitations, resulting in smaller form factor, power-efficient transceivers. In this work, biologically inspired, and afterwards motivated by the physical properties of acoustic propagation in soil, a scheme for wireless data communication through soil employing acoustic waves is presented. Experimental results illustrate the system capability of sending application-specific data, ranging from sensor readings to low-resolution images, over distances exceeding 30 m through soil.

3pSPa11. Measuring speech perception with recovered envelope cues using the peripheral auditory model. Nursadul Mamun (EE, Univ. of Texas, Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, nursad49@gmail.com), Khadija Akter (ETE, Chittagong Univ. of Eng. and Technol., Chittagong, Bangladesh), Hussnain Ali, and John H. L. Hansen (EE, Univ. of Texas, Dallas, Richardson, TX)

Speech perception refers to how understandable speech produced by a speaker would be by a listener. The human auditory system usually interprets this information using both envelope (ENV) and temporal fine structure (TFS) cues. While ENV is sufficient for understanding speech in quiet, TFS cues are necessary for speech segregation in noisy conditions. In general, ENV can be recovered from the TFS (known as recovered ENV); however, the degree of ENV recovery and its significance on speech perception are not clearly known/understood. In order to systematically assess the relative contribution of the recovered ENV for speech perception, this study proposes a new speech perception metric. The proposed metric employs a phenomenological model of the auditory periphery developed by Zilany and colleagues (J. Acoust. Soc. Am. 126, 283–286, 2014) to simulate the responses of the auditory nerve fibers to both original and recovered ENV cues. The performance of the proposed metric was evaluated under different types of noise (both steady-state and fluctuating noise), as well as several classes of distortion (e.g., peak-clipping, center-clipping, and phase jitter). Finally, to validate the proposed metric, the predicted scores were compared with subjective evaluation scores from behavioral studies. The proposed metric indicates a statistically significant correlation for all cases and accounts for a wider dynamic range compared to the existing metrics.

3pSPa12. Bat-inspired dynamic features and factors that modulate their impact on speech recognition. Alexander Hsu, Jin-Ping Han, Xiaodong Cui, Kartik Audhkhasi (IBM T. J. Watson Res. Ctr., 1101 Kitchawan Rd., Yorktown Heights, NY, Alexander.Hsu@ibm.com), Anupam Kumar Gupta (Virginia Tech, Blacksburg, VA), Joseph Sutlive (Virginia Tech, Roanoke, VA), Tabassum Ahmed, and Rolf Müller (Virginia Tech, Blacksburg, VA)

One of the most serious remaining challenges in speech recognition is dealing with corruption of speech signal by other nuisance speech ("babble"). A promising approach to solving the problem of separating the signal of interest from the detractors is to inject direction dependent signatures into all signals received, which has been realized by bat-inspired biomimetic pinna—dynamic periphery. Changing the shape of a biomimetic pinna during the recordings introduces substantial time-variant signatures into speech signals. To investigate the utility of these signatures, we have used bioinspired signal representations (cochleagram and spikegram) as input for speech classifiers based on Gaussian mixture models (GMM) and hidden Markov models (HMM). The speech samples used were obtained from open source databases: spoken digits and alphabets from Carnegie Mellon University were mixed with babble or noise samples from Columbia University. Since the time-variant signatures were found to depend strongly on the direction of the sound source, we attempted to include datasets from different directions for training and testing to feed into the classifiers. The results indicate that dynamic periphery can substantially improve recognition and that these effects depend on the signal representation as well as the angular composition of the training dataset.

3pSPa13. Is explicit formant encoding useful for speech perception with cochlear implants? Juliana N. Saba (Univ. of Texas at Dallas, 800 W Campbell Rd., Pflugerville, TX 75080, juliana.saba@utdallas.edu), Hussnain Ali, Colin Brochtrup, and John H. L. Hansen (Univ. of Texas at Dallas, Richardson, TX)

Earlier generations of cochlear implant (CI) sound processing strategies incorporated explicit encoding of formant frequencies. Many of these CI strategies have yielded higher intelligibility outcomes for CI users due to the pure spectral-based approaches used in contemporary processor today, such as: CIS, ACE, Hi-Res, etc. The aim of the study has been to assess intelligibility due to possible loss in formant frequency encoding and its influence on channel selection in diverse acoustic conditions. Formant accuracy from four computer-aided estimation techniques are compared against hand-marked frequencies derived from phoneme-level transcription with sentences from the IEEE and AzBio databases in three noise types at different SNRs: (a) babble, (b) speech-shaped, and (c) noise and reverb combination. CI speech intelligibility outcomes of the proposed strategy using explicit formant frequency encoding was compared to ACE, the standard energy-based '*n*-of-*m*' approach. A relationship between the estimation techniques and differences in channel selection was systematically assessed. Results suggest that minor loss of formant precision does not yield statistically significant differences in resulting speech intelligibility. However, availability of accurate formant cues may help with intelligibility gains in challenging noise such as babble.

3pSPa14. The CCi-MOBILE Vocoder. Hussnain Ali (Elec. Eng., The Univ. of Texas at Dallas, 800 W Campbell Rd., EC33, Richardson, TX 75080, hussnain@ieee.org), Nursadul Mamun (Elec. Eng., The Univ. of Texas at Dallas, Dallas, TX), Avamarie Bruggeman, Ram Charan M. Chandra Shekar (Elec. Eng., The Univ. of Texas at Dallas, Richardson, TX), Juliana N. Saba (Elec. Eng., The Univ. of Texas at Dallas, Pflugerville, TX), and John H. L. Hansen (Elec. Eng., The Univ. of Texas at Dallas, Richardson, TX)

The UT-Dallas Costakis Cochlear Implant Mobile (CCi-MOBILE) research platform enables experimental research with hearing-aid (HA) and cochlear-implant (CI) devices. The platform substitutes the clinical sound processor (CI/HA) with an Android smartphone/tablet or a PC as a computing hardware to run custom speech processing algorithms. The flexibility offered by such a setup enables researchers to provide custom-designed electric and acoustic stimuli (EAS) to CIs and HAs bilaterally in a time-synchronized manner. With this bimodal (electric + acoustic) capability, the platform can be used to undertake studies with individuals who either use a CI in one ear and a HA in the contralateral ear, or use hybrid implants in one or both ears. The CCi-MOBILE software suite includes a range of research applications which address different experimental needs. In the present work, a real-time VOCODER is incorporated in the CCi-MOBILE platform to facilitate research involving acoustic simulations of CIs. Two traditional flavors of VOCODER processing are implemented, namely noise-band and sine-wave vocoding. Flexibility is provided to modify processing parameters such as number of channels, frequency band-widths, filter attributes, etc. This flexibility combined with the ability to conduct long-term studies beyond the laboratory in diverse naturalistic acoustic environments will help advance research in psychoacoustics.

Session 3pSPb

Signal Processing in Acoustics: Geometric Signal Processing in Acoustics

Ananya Sen Gupta, Cochair
Electrical and Computer Engineering, University of Iowa, 4016 Seamans Center for the Engineering Arts and Sciences, Iowa City, IA 52242

Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Chair's Introduction—1:00

Invited Papers

1:05

3pSPb1. Implications of Riemannian geometry in underwater acoustics. Steven I. Finette (Acoust. Div., Code 7160, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Signal processing involving source localization and detection in underwater acoustics often depends on comparing covariance or cross-spectral density matrices (CSDMs) estimated for data, replica or noise source data from passively sensed acoustic fields incident on sensor arrays. Such comparisons often depend on a measure of similarity between matrix pairs and this typically involves a Euclidean metric. While Euclidean-based metrics are ubiquitous, convenient and useful, they are not fundamental to this task. By exploiting the facts that a CSDM is Hermitian and positive semi-definite, one can interpret such matrices as points constrained to a Riemannian, rather than a Euclidean manifold, with the implication that similarity between matrix pairs should be measured using a metric consistent with the manifold's intrinsic structure. This geometric interpretation leads to alternative matched-field processors for source localization involving the Riemannian distance as a measure of similarity and can be used to solve this inverse problem. Some implications of this non-Euclidean approach in underwater acoustics, as well as a possible extension to source detection are discussed. [Work supported by the Office of Naval Research.]

1:25

3pSPb2. The Biquaternionic acoustic wave equation. Roger M. Oba (Acoust. Div., Code 7167, Naval Res. Lab., 4555 Overlook Ave. SW, Acoust. Div., Code 7167, Washington, DC 20375, roger.oba@nrl.navy.mil)

Equations for irrotational, linearized Euler fluid, where the mass continuity equation incorporates a linear equation of state, define linear acoustics that can be recast in terms of complexified quaternions, the biquaternions, which satisfy Cauchy-Fueter regularity equations. This gives first order partial differential equations from the four space-time dimensions to the four (complex) dimensional velocity-pressure space, with time and pressure in the biquaternionic scalar parts. The biquaternionic planar exponential provides the basis to elementary solutions as biquaternionic functions of a single biquaternionic variable. Analysis of this form shows significant geometrical algebraic properties including imaginary units for time distinct from those for space. The single biquaternion acoustic field at each point of space-time allows the propagation four complex coefficients in a single geometric algebraic structure, suitable for three or four dimensional Fourier analysis. (Work supported by the U.S. Office of Naval Research.)

1:45

3pSPb3. Fermat's principle, Fresnel zones, and ray methods for high-frequency signatures of elastic cylinders in water. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Geometric methods are useful for understanding the magnitude and timing of high-frequency echoes from elastic objects in water. This presentation emphasizes, but is not limited to, elastic contributions associated with leaky waves on circular cylinders. The regions of strong coupling between the incident acoustic wave and guided leaky waves can be predicted using Fermat's principle while the size of the associated Fresnel zones are useful for understanding signal magnitudes [P. L. Marston, J. Acoust. Soc. Am. 102, 1628–1638 (1997); F. J. Blonigen and P. L. Marston, 110, 1764–1769 (2001)]. Ray theory predictions are supported by measurements with tilted cylinders [S. F. Morse, P. L. Marston, and G. Kaduchak, J. Acoust. Soc. Am. 103, 785–794 (1998); K. Gipson and P. L. Marston, J. Acoust. Soc. Am. 106, 1673–1680 (1999); F. J. Blonigen and P. L. Marston, J. Acoust. Soc. Am. 112, 528–536 (2002)]. Related applications include signatures and images of tilted cylinders near flat surfaces [D. S. Plotnick and P. L. Marston, J. Acoust. Soc. Am. 140, 1525–1536 (2016)], time-frequency signatures [S. F. Morse and P. L. Marston, J. Acoust. Soc. Am. 111, 1289–1294 (2002)], and signatures for other targets with corners. [Work supported by ONR.]

2:05

3pSPb4. Adaptive integration for Capon-type beamformers by exploring Riemannian geometry on covariance matrices. Magnus L. Nordenvaad (Marine Systems, Swedish Defence Res. Agency (FOI), Gullfossgatan 6, Stockholm, Sweden 16490, Sweden, maglun@foi.se)

Adaptive, or Capon beamformers, offers several advantages compared to standard alternatives in underwater sensing. Furthermore, for hull mounted sensor arrays, adaptive beamformers is a convenient way of mitigating platform induced interference. The underlying difficulty in implementing a data-adaptive approach is how to obtain an accurate estimate of the data covariance matrix, R. The combination of high-dimensionality and time-variation renders significant challenges in maintaining accurate covariance estimates. A common way to approach this issue is to update the covariance matrix using a weighted Euclidean average, R_{new} = a R_{old} + (1-a) R_{inst}, where R_{inst} is some instantaneous measurement. Meanwhile, the time-variation naturally form a trajectory on the covariance matrix manifold. With this in mind, it makes much more sense to smooth the trajectory on the space of covariance matrices using proper metrics. The purpose of this paper is to take the geometry of covariance matrices into account while performing adequate covariance matrix estimation. Specifically, we propose a simple and elegant algorithm to track the covariance based on an update through means on Riemannian manifolds. This solution is then incorporated into an adaptive beamformer, and initial evaluations based on both simulated and real data show slight improvements compared to standard approaches.

Contributed Papers

2:25

3pSPb5. Fast two-dimensional censored mean level detector-statistics constant false alarm rate method for multi-beam seafloor terrain detection. Jiaqi Wang and Haisen Li (Harbin Eng. University, China, 145th, Str. Hantong, Harbin, Heilongjiang Province 150001, China, wangjiaqi@hrbeu.edu.cn)

In order to solve the problem of false terrain caused by environmental interference and “tunneling effect” in conventional multi-beam seafloor terrain detection, this paper addresses a seafloor topography detection method based on fast two-dimensional (2D) CMLD-CFAR (censored mean level detector-Statistics Constant False Alarm Rate). A cross-sliding window is used in this method. The target occlusion phenomenon which occurs in multi-target environments can be eliminated by censoring the large cells of the reference cells, and the rest reference cells are used to calculate the local threshold. The conventional 2D CMLD-CFAR methods need to estimate the background clutter power level for every pixel, which costs a lot of computational load. This paper proposes a fast algorithm with a global threshold to reduce the computational load. Only the regions of interest (ROI) which are selected by a global threshold will be detected by 2D CMLD-CFAR method, and the rest pixels are distinguished as clutters directly. The proposed method has been evaluated by the real multi-beam experimental data of which the background follows a distribution of Exponential. The results show that the novel method can effectively solve the problem of false terrain in multi-beam terrain survey and has a high detection accuracy.

2:40

3pSPb6. Representing sonar target features using Braid geometry. 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)

Acoustic color features that specify a sonar target in the time-frequency spectral domain may be broadly classified into two groups: (i) elastic wave features that depend on the material composition of the target, and (ii) wave features that depend on the specific geometry of the sonar target. Typically these two types of target features overlap intricately in the acoustic color domain that renders them difficult to separate using traditional signal processing techniques. Furthermore, field experimental conditions such as target orientation, proud, buried or semi-buried state of the target as well as sediment characteristics, influence the relative overlap and morphology of these acoustic color features. A key bottleneck to robust target feature representation and subsequent classification is how these potentially overlapping features may change in the field from what is measured under laboratory conditions or observed through model-based simulations. We will present some relevant results from our ongoing work in disentangling sonar features using a combination of geometric signal processing, graph-based informatics techniques and well-known physical models. Specifically, we will employ geometry of braid manifolds as a possible basis for representing the physical phenomena influencing each group of target features. We will also present recent results from graph-based informatics to show how braid encoding across multiple experimental conditions can be employed to determine and disentangle overlapped target features.

2:55

3pSPb7. Harnessing geometric techniques for robust real-time estimation of shallow water acoustic channels. Ananya Sen Gupta and Ryan McCarthy (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 delay spread, which poses a fundamental roadblock to real-time channel estimation, particularly under moderate to rough sea conditions. This talk will summarize recent research on multiple representations of the shallow water acoustic channel and present the relative merits of each channel representation in localizing different channel effects. In particular, we will examine the efficacy of employing geometric signal processing techniques to adaptively track the shallow water acoustic channel in each representation. Results based on experimental field data as well as channel simulations under varying environmental conditions will be presented.

3:10–3:30 Panel Discussion

ASA Plenary Session and Awards Ceremony/CAA Annual General Meeting

Lily M. Wang, Cochair
President, Acoustical Society of America

Jérémie Voix, Cochair
President, Canadian Acoustical Association

Traditional Lkwungen Territory Welcome

Acoustical Society of America Plenary Session and Awards Ceremony

3:45 p.m. to 4:45 p.m.

Presentation of ASA Fellowship Certificates

Ahmed A. Al-Jumaily – For contributions to biomedical applications of acoustics and vibrations

Richard D. Costley, Jr. – For contributions to the analysis of battlefield acoustics and the acoustical properties of structures

Jan Dettmer – For contributions to inversion methods and uncertainty quantification applied to geoacoustics

Joseph R. Gladden, III – For service to and leadership in the field of physical acoustics

Dorian S. Houser – For advancing the understanding of the impacts of anthropogenic noise on marine mammals

Brian F. Katz – For contributions to measure teechniques and spatial hearing for auditory virtual reality

Colleen Reichmuth – For contributions to pinniped acoustics

Gary P. Scavone – For contributions to the analysis and modeling of musical instruments

Eleanor P. Stride – For contributions to the modelling, development, and manufacturing of acoustically responsive biomaterials

David S. Woolworth – For contributions to general architectural acoustics and service to the Society

Catherine L. Rogers – For contributions to speech communication through service, mentoring, and scholarship

Presentation of Acoustical Society Awards

Wallace Clement Sabine Medal to Michael Vorländer

Canadian Acoustical Association Annual General Meeting

4:45 p.m. to 5:30 p.m.

Presentation of Canadian Acoustical Association Awards

Shaw Postdoctoral Prize in Acoustics to Olivier Valentin (École de technologie supérieure, Montréal) for "EARtrode, A Wireless In-ear Custom-fitted intelligent Brain Computer Interface"

Bell Student Prize in Speech Communication and Hearing to Megan Keough (University of British Columbia, Vancouver) for "Reafferent Feedback and Aerotactile Integration"

Fessenden Student Prize in Underwater Acoustics to Josee Belcourt (University of Victoria) for "Bayesian Geoacoustic Inversion of Seabed Reflection Data at the New England Mud Patch"

Bregman Student Prize in Psychological Acoustics to Sean Gilmore (Ryerson University, Toronto) for "Feeling the Beat: An Investigation into Tactile Beat Perception"

Northwood Student Prize in Architectural and Room Acousticsto Magdalenn Bahour (Ryerson University, Toronto) for "Living Wall and Acoustic Comfort - A Case Study"

Canada-Wide Science Fair Award to Zachary Trefler (Waterloo Collegiate Institute) for "VoiceShield: Teaching Computers to Distinguish Real Data From Fake"

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

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

This workshop for Victoria Girl Guides consists of hand-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.

OPEN MEETINGS OF TECHNICAL COMMITTEES/SPECIALTY GROUPS

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, 6 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Rattenbury A/B (FE)
Acoustical Oceanography	7:30 p.m.	Esquimalt (VCC)
Animal Bioacoustics	7:30 p.m.	Oak Bay 1/2 (VCC)
Architectural Acoustics	7:30 p.m.	Theater (VCC)
Physical Acoustics	7:30 p.m.	Colwood 1/2 (VCC)
Psychological and Physiological Acoustics	7:30 p.m.	Salon B (VCC)
Speech Communication	7:30 p.m.	Salon A (VCC)
Structural Acoustics and Vibration	8:00 p.m.	Saanich 1/2 (VCC)

Committees meeting on Wednesday, 7 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Sidney
Signal Processing in Acoustics	7:30 p.m.	Colwood 1/2

Committees meeting on Thursday, 8 November

Committee	Start Time	Room
Computational Acoustics	4:30 p.m.	Esquimalt (VCC)
Musical Acoustics	7:30 p.m.	Crystal Ballroom (FE)
Noise	7:30 p.m.	Shaughnessy (FE)
Underwater Acoustics	7:30 p.m.	Rattenbury A/B (FE)

WALLACE CLEMENT SABINE AWARD OF THE ACOUSTICAL SOCIETY OF AMERICA

Michael Vorländer

2018

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

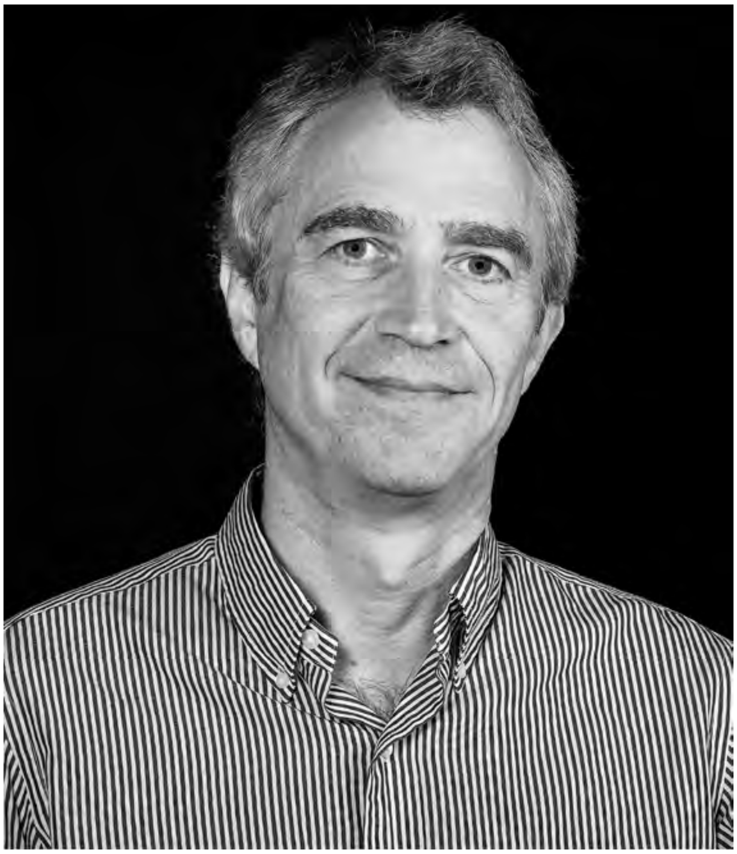
Vern O. Knudsen	1957	A. Harold Marshall	1995
Floyd R. Watson	1959	Russell Johnson	1997
Leo L. Beranek	1961	Alfred C. C. Warnock	2002
Erwin Meyer	1964	William J. Cavanaugh	2006
Hale J. Sabine	1968	John S. Bradley	2008
Lothar W. Cremer	1974	J. Christopher Jaffe	2011
Cyril M. Harris	1979	Ning Xiang	2014
Thomas D. Northwood	1982	David Griesinger	2017
Richard V. Waterhouse	1990		

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 MICHAEL VORLÄNDER

. . . for contributions to room-acoustic simulations and virtual auditory displays

7 NOVEMBER 2018 • VICTORIA, CANADA

Michael Vorländer was born in Duisburg, Germany. At age 17 he was interested in basketball and playing jazz but during his high school years he became intrigued by physics, mathematics, and astronomy. Also, during this time Michael met his wife Angelika while both were still in high school. He began his university studies in physics at the University of Dortmund in 1978. During that period he was accepted as a student intern at the Institut für Bauphysik (IfB) which led him to think of combining his own interests with those of his father's in physics and architecture. For the IfB Michael conducted field measurements on sound insulation and programmed protocols on a HP 41C PRM calculator. This internship was the origin of his journey into acoustics.

In 1980, Angelika was accepted to study Art and Design in Aachen, which led Michael to transfer to the RWTH Aachen University after his 'Vordiplom' (prerequisite for a full Diploma). At the RWTH Aachen, Michael took solid-state physics and astronomy classes, along with courses in acoustics taught by Professor Heinrich Kuttruff. In his Master's thesis research, Michael calculated Michael Barron's lateral energy ratio with image sources in performance venues on an Apple II. This research resulted in his first peer-reviewed journal paper co-authored with Professor Kuttruff in ACUSTICA 58, 118-129, 1985.

Michael and Angelika married in 1984 and he began his doctoral research with Professor Kuttruff in 1985. He completed his doctoral dissertation in 1989 with three additional journal publications. Among them was his proposal for a hybrid ray-tracing and image source method which led to great progress in efficient computer simulations in room-acoustics (Journal of the Acoustical Society of America 86, 172-178, 1989).

Following his doctoral degree, Dr. Vorländer accepted a postdoctoral position at the PTB Braunschweig, the National Institute of Metrology in Germany. Here Michael shifted his research focus to audiological measurements, binaural technology and microphone calibration. Two years later, in 1991, Michael was appointed the Head of the laboratory of building acoustics at the PTB, which was responsible for room and building acoustics, sound emission and absorption measurement along with microphone calibration in reverberation chambers.

While working for the PTB, Dr. Vorländer completed the 'habilitation' (the inauguration dissertation) in 1994 at the Technical University of Dresden. This led Michael to be named a Professor at the RWTH Aachen in 1996, succeeding Professor Kuttruff, as the head of the Institute of Technical Acoustics (ITA). Professor Vorländer expanded his research to a wide variety of high-impact topics including room-acoustics and virtual auditory reality. He leads an extremely productive research team that, so far, has amassed over 100 peer-reviewed journal articles.

Michael has authored a monograph entitled *Auralization* (Springer, 2007), which has become an influential volume in the architectural acoustics field and the virtual auditory reality community, and has been cited over 540 times. He has also contributed to 15 book chapters including his latest chapter 'Auralization' for the *Architectural Acoustics Handbook* (J. Ross Publishing 2017, Ed. N. Xiang). In another milestone contribution to architectural acoustics, Michael and Dr. E. Mommertz, defined the scattering coefficient of interior surfaces in enclosures along with a methodology to experimentally measure it (Journal of Applied Acoustics 60, 187-199, 2000). This contribution made the materials and devices of architectural acoustics rigorously quantifiable, which considerably advanced the room-acoustics and computer-simulation field. Since then, this work was also adopted as an International Standard (ISO 17497-1, 2004).

Dr. Vorländer is a Fellow of the Acoustical Society of America (ASA). He has served as a member and chair of the Society's Committee on International Research and Education (CIRE) and he was recently elected a Member of the Executive Council. Michael has served as a JASA Associate Editor for Architectural Acoustics since 2013. He has presented many invited papers as well as organized or co-organized a number of architectural acoustics sessions at ASA meetings.

Dr. Vorländer has played significant roles at the scientific and administrative leadership levels in German, European, and international communities. He served as Vice President and President of the German Acoustical Society (2013-2016, 2016-2019), Vice President and President of the European Acoustics Association (2004-2007, 2007-2010), President of the International Commission for Acoustics (2010-2013), and Editor-in-Chief of the journal Acta Acustica united with Acustica (1998-2003). His international meeting service includes Technical Co-chair of the International Congress on Acoustics (ICA) in 2007, General Co-chair of Acoustics ’08 Paris, and General Chair of the ICA congress to be held in 2019 in Aachen. He was awarded the R. W. B. Stephens Medal of the Institute of Acoustics in 2005 and the 2014 EAA Award for contributions to the promotion of acoustics in Europe.

Recreationally, he enjoys playing drums and is often seen at the Jam Sessions at ASA meetings. Even though there is no basketball for Michael these days, he still plays soccer with his ITA colleagues.

There is no doubt that Dr. Michael Vorländer has made distinguished contributions to acoustics research, publications, and leadership in acoustical societies. We are delighted that the Acoustical Society of America recognizes and honors Dr. Michael Vorländer with this prestigious Wallace Clement Sabine Medal.

NING XIANG
LAURI SAVIOJA

Session 4aAA

Architectural Acoustics, Engineering Acoustics, Signal Processing in Acoustics, and Noise: Microphone Array Applications in Room Acoustics

Gary W. Elko, Cochair
mh Acoustics, 25A Summit Ave., Summit, NJ 07901

Michael Vorländer, Cochair
ITA, RWTH Aachen University, Kopernikusstr. 5, Aachen 52056, Germany

Chair's Introduction—7:45

Invited Papers

7:50

4aAA1. Practical open-concentric spherical microphone array design. Mark R. Thomas (Sound Technol. Res., Dolby Labs., 1275 Market St., San Francisco, CA 94103, mark.r.thomas@ieee.org)

The problem of higher order sound field capture is considered. First-order microphone arrays, such as tetrahedral A-format designs, are commonplace, while interest remains in the increased spatial resolution delivered by higher order arrays. Such arrays typically consist of pressure microphones mounted on a solid spherical baffle, with which higher-order spatial components are estimated algorithmically. This produces a design trade-off, with small arrays being preferred for spatial aliasing performance and large arrays being preferred for reduced amplification of microphone capsule noise at low frequencies. A practical open sphere design is proposed that contains microphones mounted at multiple radii to fulfill both criteria. Coupled with the use of cardioid microphones, such an array captures higher order soundfields over a wider audio bandwidth than baffled spherical designs of fixed radius.

8:10

4aAA2. Self-calibration of dual, closed surface microphone arrays. Earl G. Williams (Acoust. Div., Code 7106, Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, earl.williams@nrl.navy.mil) and William A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

With the recent availability of very inexpensive, miniature digital MEMS microphones, the construction of large count microphone arrays is financially within the reach of the basic researcher. Using various mathematical techniques such as the Helmholtz-Kirchhoff integral and the equivalent source method (ESM), dual surface microphone arrays provide the ability to separate incident and scattered fields instantaneously in time. When these arrays form a closed surface, this separation is extremely accurate. However, one of the concerns with any microphone array is the calibration of the individual elements, especially near the resonance frequency of the diaphragm. In this paper, a self-calibration technique is proposed, based on ESM, that uses external loudspeakers in a simple laboratory facility not required to be anechoic. The approach is tested numerically on a dual surface, 256 element pair spherical microphone array, and uses an algorithm for the minimization of a cost function based on the extinction of the scattered field when the unknown individual calibration coefficients of the array elements are varied. Use of a Burton-Miller approach to model the ESM sources of the scattered field improves dramatically the accuracy at specific frequencies. The problematic interior Dirichlet eigenfrequencies are addressed. [Work supported by the Office of Naval Research.]

8:30

4aAA3. Characterizing acoustic environments using spherical loudspeaker and microphone arrays. Hannes Gamper, Keith Godin, Nikunj Raghuvanshi, and Ivan J. Tashev (Res., Microsoft, One Microsoft Way, Redmond, WA 98052, hannes.gamper@microsoft.com)

Room acoustic parameters, including the reverberation time (T60) and the direct-to-reverberant ratio (DRR), quantitatively describe the behaviour of an acoustic environment. These parameters are typically derived from an acoustic impulse response (AIR) obtained using an omnidirectional sound source and an omnidirectional receiver. However, the acoustic interaction between a source with a complex radiation pattern, for example, a human talker, and a receiver with a complex directivity pattern, for example, a human listener, cannot be accurately described by an omnidirectional AIR or the acoustic parameters derived from it. Here, we propose characterizing acoustic environments while taking the source and receiver directionality into account. A spherical loudspeaker array allows modelling a source with an arbitrary radiation pattern. Analogously, a spherical microphone can be used to describe a receiver with an arbitrary directivity pattern. By combining spherical loudspeaker and microphone arrays, a multiple-input multiple-output (MIMO) AIR of an acoustic environment can be measured to simulate sources and receivers with arbitrary directivities and estimate acoustic parameters accordingly.

8:50

4aAA4. The room impulse response in time, frequency, and space: Mapping spatial energy using spherical array beamforming techniques. 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)

The auditory perception of rooms is a multi-dimensional problem. Our hearing system interprets time, frequency, and spatial information from arriving room reflections, but traditionally, only the time and frequency domains are considered in room acoustic metric and objective sound field analyses. This work aims to develop spatial visualizations of the energy in a room impulse response (RIR). With a spherical microphone array, a room's energy can be mapped in full three-dimensions. First, beamforming techniques are used to generate a set of directional RIRs from the spherical microphone array measurement. This set of directional RIRs is analogous to using a microphone with a directional beam pattern response, oriented individually at all points around a sphere. Then, these directional or beam RIRs are time windowed and band-pass filtered to create spatial energy maps of the room. Comparisons between a plane-wave beam pattern and a Dolph-Chebyshev beam pattern will be demonstrated in the context of RIR beamforming. As well, different strategies for normalizing peak energy amplitudes to either the direct sound or a spherical spreading condition will be compared. With these considerations, final results of these spatial energy visualizations and directional RIR animations will be demonstrated. [Work supported by NSF Award 1302741.]

9:10

4aAA5. Model-based direction of arrival estimations for sound sources using a spherical microphone array. Christopher R. Landschoot (Architectural Acoustics, Rensselaer Polytechnic Inst., 54 West Main St., Clifton Springs, NY 14432, crlandschoot@gmail.com), Jonathan Mathews, Jonas Braasch, and Ning Xiang (Architectural Acoustics, Rensselaer Polytechnic Inst., Troy, NY)

In many room acoustics and noise control applications, it is often challenging to identify the directions of arrivals (DoAs) of incoming sound sources. This work seeks to solve this problem reliably by beamforming, or spatially filtering, incoming sound data with a spherical microphone array via a probabilistic method. When estimating the DoA, the signal under consideration may contain one or multiple concurrent sound sources originating from different directions. This leads to a two-tiered challenge of first identifying the correct number of sources, followed by determining the directional information of each source. To this end, a probabilistic method of model-based Bayesian analysis is leveraged. This entails generating analytic models of the experimental data, individually defined by a specific number of sound sources and their locations in physical space, and evaluating each model to fit the measured data. Through this process, the number of sources is first estimated, and then the DoA information of those sources is extracted from the model being the most concise to fit the experimental data. This paper will present the analytic models, the Bayesian formulation, and preliminary results to demonstrate the potential usefulness of this model-based Bayesian analysis for complex noise environments with potentially multiple concurrent sources.

9:30

4aAA6. Iterative echo labeling technique for room geometry estimation. Soo Yeon Park and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 34141, South Korea, parksean210@kaist.ac.kr)

Room geometry estimation using acoustic room impulse responses (RIRs) has been tackled in various ways. Most of them detect wall locations using the time of arrival (TOA) of echoes between multiple microphones and loudspeakers. To combine TOA information from multiple room impulse responses, echoes from the same walls should be collected and processed together. When multiple microphones are widely distributed in a room, however, it is hard to identify walls responsible for the generation of each echo. In this work, an iterative update algorithm is proposed for resolving the echo labeling problem with low computational complexity. Unlike conventional algorithms utilizing the rank property of Euclidean distance matrix, the proposed algorithm utilizes the property of convex hull formed by ellipses representing individual TOAs. The location of the reflectors are sequentially identified from common tangent lines constituting the convex hull, and the search result is iteratively updated by inspecting later echoes in time. In addition, a termination criterion is also developed in order to enable the geometry estimation without the prior knowledge of the number of the reflectors.

9:50

4aAA7. Nonsingular EB-ESPRIT for the localization of early reflections in a room. Byeongho Jo and Jung-Woo Choi (Elec. Eng., Korea Adv. Inst. of Sci. and Technol., #2103 N24 LG Innovation Hall KAIST, 291 Daehak-ro Yuseong-gu, Daejeon 34141, South Korea, byeongho@kaist.ac.kr)

The eigenbeam-ESPRIT (EB-ESPRIT) can directly identify the direction of arrivals (DoAs) of sound sources through the parametric estimation process. This parametric estimation is beneficial for identifying directions of early echoes in an enclosed space, because it does not need an extra peak searching step for finding dominant peak locations from a beamforming power map. Nevertheless, the use of EB-ESPRIT for echo localization has been hindered due to its ill-conditioning problem for sources positioned near the equator of the spherical coordinates. Room reflections in a reverberant room can impinge from any direction, and hence, the ill-conditioning is highly likely to occur. In this work, a nonsingular EB-ESPRIT technique is utilized for the echo localization without the ill-conditioning problem. The technique is based on new recurrence relations of spherical harmonics that express DoA parameters in form of sinusoidal functions. Therefore, the ill-conditioning due to the use of arctangent function can be avoided. The performance is demonstrated using the room impulse responses simulated and measured for a rigid spherical microphone array. Frequency smoothing is applied to localize echoes that are highly coherent to each other, and results demonstrate that accurate localization of echoes is possible irrespective of their incidence angles.

10:10–10:25 Break

10:25

4aAA8. Analysis of sound field isotropy based on directional energy decay curves. Marco Berzborn and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, marco.berzborn@akustik.rwth-aachen.de)

The analysis of the diffuseness of sound fields is of great interest in room acoustic applications ranging from the analysis of concert venues to reverberation room design and calibration. However, a standardized robust diffuseness estimation method is currently lacking. The fundamental definition of the diffuse sound field is that it is isotropic—requiring the sound field to be composed of infinitely many sound waves from uncorrelated sources with directions of arrival uniformly distributed over a sphere. Due to their symmetry, spherical microphone arrays are especially favorable for the analysis of the isotropy requirement. In this work, we propose the directional energy decay curve which we calculate from a directional room impulse response captured with a spherical microphone array. Further, we show how the analysis of the spatial variations of the directional energy decay curve enables examining the isotropy of sound fields in rooms. Finally, we present a simulation study of multiple room configurations with varying degrees of sound field diffuseness.

10:45

4aAA9. Beamforming loudspeaker array in a multi-layered configuration. Jeong-Guon Ih (Mech. Eng., KAIST, 373-1 Guseong-Dong, Yuseong-Gu, Daejeon 305-701, South Korea, J.G.Ih@kaist.ac.kr) and Wan-Ho Cho (KRISS, Daejeon, South Korea)

A planar, broadside acoustic array with uniform loudspeaker spacing is often adopted for generating a highly directional acoustic beam, but the performance in radiated power and spatial bandwidth should always compromise with the transverse size limit. To achieve a high-power, directional sound in a compact size, a multi-layered loudspeaker array configured in both broadside and end-fire types is devised. This design concept is tested with an array of 3 layers with 0.2 m spacing. Each layer contains 7 loudspeakers in a centrally symmetric arrangement on a plane with the longest dimension of 0.6 m. In each layer, the separation distance between loudspeakers is uniformly set as 0.2 m. The optimal beamforming method is employed for calculating the control filter. The compensation for the performance degradation due to self-scattering effect is also included in the control filter, for which the additional phase delay due to scattering is estimated from the simulated response of the array system. The maximum sound level at 1 m distance from the array center is achieved as 145 dB at 1.5 kHz, and, at 30 m position, the level decrease of 26–30 dB with the directivity index of 15 dB is observed for 1–5 kHz.

11:05

4aAA10. Modal reconstruction of the sound field in a room at low frequencies. Efren Fernandez-Grande (Acoust. Technol., DTU - Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk)

Capturing the spatio-temporal (or spatio-spectral) properties of the sound field in a room is valuable for its characterization. To measure the full spatio-spectral properties within the volume of the room, i.e., the frequency response functions at all points, a very large number of measurements is required. Even at low frequencies, sampling the three-dimensional space enclosed by a room is challenging. It is therefore of interest to employ sound field reconstruction methods to predict (interpolate/extrapolate) the sound field over the volume of the room, based on a limited number of measurements. The aim of this study is to develop a sound field reconstruction method that exploits the modal structure of the sound field in a room, to reconstruct the sound field at low frequencies over the entire volume of the space. The proposed methodology makes use of the fact that the spatial structure of the acoustic field in a room is inherently block-sparse and can be extracted from spatially distributed measurements. The modeshapes are represented as the superposition of a small number of plane waves, and the modal basis is used to reconstruct the sound field.

11:25

4aAA11. Postfiltering based on the coherent-to-diffuse power ratio. Michael Günther, Andreas Brendel, and Walter Kellermann (Chair of Multimedia Communications and Signal Processing, Friedrich-Alexander-Universität Erlangen-Nürnberg, Cauerstr. 7, Erlangen 91058, Germany, michael.guenther@fau.de)

In microphone array processing, the availability of multiple sensors observing an acoustic scene allows the extraction of additional spatial sound field properties. The coherent-to-diffuse power ratio (CDR) is a signal-dependent quantity which relates the coherent, typically desired, component and the diffuse, typically undesired, component of a recorded signal and can be employed in different signal enhancement tasks such as dereverberation and noise suppression. Given suitable models for the coherent and diffuse signal components, the complex spatial coherence function of the observed sound field facilitates an efficient estimation of the CDR. Several unbiased CDR estimators can be derived which differ in the sensitivity to estimation errors and the amount of incorporated prior information, e.g., the direction of arrival (DOA) of a desired point-like source. We review popular DOA-dependent and DOA-independent CDR estimators and investigate their efficacy in single-channel postfiltering when multiple point-like acoustic sources, i.e., one target source and at least one coherent interferer, are present.

Contributed Papers

11:45

4aAA12. Measurement of room spatial correlation functions using spherical microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoust., Fairfax, Vermont)

The study of acoustic spatial correlation functions has been ongoing since early analytical development and experiments were done by Farran and Hills in 1952. The measurement of sound diffuseness using spatial correlation functions has been discussed previously by the authors in 2009. It has been conjectured that audio rendering of the spatial impression of a soundfield is related to the degree of correlation between the microphones used to record the soundfield. The correlation between the recorded channels is related to the diffuseness of the sound field. Many different models of diffuse fields have been described and are based on a definition that involves incoherent sound coming uniformly from all directions. Schroeder surmised that as the modal density exceeds three overlapping modes in one modal bandwidth that the random uniform model can be utilized in practice.

In order to obtain measured room spatial correlation functions one has to ensemble average either multiple source or receiver positions. With eigenbeamforming spherical microphone arrays these functions can be ensemble averaged by utilizing multiple steered beampatterns with at least two spaced arrays. We will show some theoretical spatial correlation functions for different beampatterns and orientations and compare these to measurements made in a real room.

12:00

4aAA13. Transient solution for the directional response at the focus of a paraboloidal reflector. John M. Cormack and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

A transient solution is presented for the directional response at the focus of a paraboloidal reflector due to an incident plane wave. The validity of the solution is associated with the Kirchhoff approximation used to determine the boundary condition on the surface of the reflector, which requires the wavelengths to be short compared with the minimum radius of curvature of the reflector. The solution at the focus due to an incident plane wave is obtained by applying the principle of reciprocity to the solution in the far field due to a point source at the focus. Both solutions are expressed as the convolution of the unit step response with the time derivative of the waveform incident on the reflector. For a point source far removed from a shallow paraboloidal reflector, it is shown that the pressure in the far field reduces to the expression obtained by Morse (*Vibration and Sound*) for transient radiation from a baffled circular piston. Results are presented illustrating the angular dependence of the reflected pressure waveforms at the focus for incident plane waves that include a rectangular pulse, an N wave, and a tone burst. [J.M.C. was supported by the ARL:UT McKinney Fellowship in Acoustics.]

THURSDAY MORNING, 8 NOVEMBER 2018 SHAUGHNESSY (FE), 8:00 A.M. TO 12:00 NOON

Session 4aAB

Animal Bioacoustics: Soundscapes, Noise, and Methods in Animal Bioacoustics

Carrie C. Wall, Chair
Cooperative Institute for Research in Environmental Sciences, University of Colorado at Boulder, 216 UCB, Boulder, CO 80309

Contributed Papers

8:00

4aAB1. An experiment of sound effects on shrimps for ecoacoustics' research. Zhengliang Cao (Shanghai Ocean Univ., 999 Hucheng Huan Rd., Shanghai 201306, China, caozhengliang@yahoo.com)

It is important to understand relationships of underwater soundscapes, marine ecology, and acoustic behaviors for refine research questions on ecoacoustics. Benthic crustaceans live in a local range and are normally used to an indication of environmental impacts, especially for the ecosystem in ocean. To get much information of interaction impacts on benthic crustaceans, this paper introduces an experiment of sound effects on shrimps in an aquatic pond. In the experiment, there is only a kind of benthic crustaceans, Penaeus prawns. Environmental and artificial noises are recorded by a passive sound monitor. The preliminary results are obtained through preprocessing data acquisition, analyzing effective phenomena, and comparing sound characteristics. It will be valuable to consider much simple cases for main questions. So data of natural rainfall noise, aerator noise and artificial noise may help to know about the acoustic ecology of shrimps in an aquaculture environment. Since the water quality is very poor in real shrimp pond, it cannot accurately obtain the relationship between the sound of shrimp and its behavioral characteristics. Further research considerations are also be prospected in this paper. [The project is funded by The National Natural Science Foundation of China (41374147) and (14111103900).]

8:15

4aAB2. A new model for underwater noise research in larval fishes: Biomedical and ecological implications. Allison Coffin (Integrative Physiol. and Neurosci., Washington State Univ., 14204 NE Salmon Creek Ave., Vancouver, WA 98686, allison.coffin@wsu.edu), Jie Xu, Dmitry Gritsenko (Mech. and Industrial Eng., Univ. of Illinois at Chicago, Chicago, IL), Kristy Lawton (Integrative Physiol. and Neurosci., Washington State Univ., Vancouver, WA), Beija Villalpando (College of Arts and Sci., Washington State Univ., Vancouver, WA), Joseph A. Sisneros (Psych., Univ. of Washington, Seattle, WA), Ashwin Bhandiwad (Psych., Univ. of Washington, Bethesda, MD), and Phillip Uribe (Integrative Physiol. and Neurosci., Washington State Univ., Vancouver, WA)

In humans, excessive noise exposure from occupational or recreational sources causes permanent hearing loss. Similarly, exposure to underwater anthropogenic noise can cause hearing loss in aquatic organisms, including fish. While fish can recover from noise-induced hearing loss, underwater noise exposure can cause behavioral changes that reduce organismal fitness. In all vertebrates, acoustic trauma can cause damage to sensory hair cells.

To better study the effects of noise on hair cells, we have developed a noise exposure system that uses broadband sound to damage hair cells of the inner ear and lateral line of larval zebrafish. Acoustic over-exposure kills hair cells in an intensity- and time-dependent manner, with maximum hair cell damage observed 72 hours after noise exposure. This time course is consistent with mammalian studies, where hair cell death occurs days to weeks after noise exposure. Other features of acoustic trauma are also conserved between zebrafish and mammals, including activation of apoptotic signaling cascades and changes in hair cell-afferent synapses. These studies demonstrate that larval zebrafish are a tractable new model for studies of noise-induced hair cell death. However, our acoustic trauma system could also be used in other species, allowing for new studies of underwater noise in larval fishes.

8:30

4aAB3. Potential changes in the communication space of humpback whale social sounds in increasing wind and vessel-dominated noise. Rebecca Dunlop (School of Veterinary Sci., Univ. of Queensland, Cetacean Ecology and Acoust. Lab, Gatton, QLD 4343, Australia, r.dunlop@uq.edu.au)

An animal communication network involves complex acoustic interactions between multiple senders, receivers, and eavesdroppers. Any reduction in communication space, due to signal masking, may have detrimental effects on their ability to obtain social information. Humpback whales use social sounds (vocal and surface-generated percussive sounds) for within-, and between-, group communication. To generate masking models, and infer communication space, changes in signal-above-noise and frequency content of received humpback whale social sounds were modelled with the combined effect of increasing background noise (wind or vessel-dominated) and distance from the source (signalling whale). Results suggest that the signaler's communication space in increasing wind-dominated noise (shallow water) was maintained out to approximately 3 km by using a Lombard response. In high wind noise (over 105 dB re 1 μPa; 12–15 knots), the vocal communication space was significantly reduced, though the increased use of surface-generated sounds likely aided in maintaining this space. In vessel-dominated noise, communication space was significantly reduced in levels exceeding 110 dB re 1 μPa (vessel within approximately 2 km), with no evidence that the signaler switched to using surface-generated sounds. These models can be updated as more information on humpback auditory capabilities becomes available.

8:45

4aAB4. Assessing risk of underwater noise impact on marine mammals throughout a new methodology. Marie Mauran, Florent Le Courtois, Marie Cachera, Yann Stéphan (HOM, Shom, 13 Rue de Châtellier, Brest 29200, France), Jérôme Spitz (Observatoire PELAGIS - UMS 3462, La Rochelle, France), and G. Bazile Kinda (HOM, Shom, Brest, France, bazile.kinda@shom.fr)

Shipping noise has been identified as a threat on underwater ecosystems. In particular, several impacts have been documented on cetaceans, e.g., communication's masking and stress increasing. To assess the risk related to such anthropogenic noise requires both quantifying the noise levels and estimating the distribution of the cetaceans' population. However, current methods evaluating the risks related to anthropogenic pressures generally rely on strong expert priors, which may be difficult to define. This presentation aims at introducing a new framework for the comparison of anthropogenic pressure levels maps and cetaceans' distribution in order to infer the risk of impact and provide management solutions. The methodology, combining simple statistical analyses and a theoretical representation tool was applied to the Bay of Biscay using shipping noise model and fin whale observation from regular surveys for the years 2012 and 2016. Relationships between cetaceans' distribution and noise levels were investigated and linked to mammals theoretical responses to pressure. A trend analysis between 2012 and 2016 was also proposed to identify the noise hotspots. The results were interpreted in both terms of ecological meaning for fin whale and conservation measures for shipping noise, according to marine policies requirements.

9:00

4aAB5. Monitoring noise levels and delphinid presence through autonomous acoustic in Rio de Janeiro coastal area. Lis Bittencourt (Programa de Pós-Graduação em Oceanografia, Rio de Janeiro State Univ., São Francisco Xavier, 540, sala 4002, bloco E, Rio de Janeiro, RJ 20550013, Brazil, lis.bitt@gmail.com), André Sousa, Tatiana Bisi, José Lailson-Brito, and Alexandre d. Azevedo (MAQUA - Laboratório de Mamíferos Aquáticos e Bioindicadores, Rio de Janeiro State Univ., Rio de Janeiro, Brazil)

In order to investigate the soundscape in a Rio de Janeiro coast site, passive acoustic monitoring was conducted during two non-consecutive weeks in the summer of 2015/2016 by deploying one SM2M + device. The equipment recorded at a 66% duty cycle with sample rate of 96 kHz and 36 dB gain. Third octave levels (TOLs) were calculated for all recordings through PAMGuide software. To search for delphinid presence, a band limited energy detector was employed using Raven 1.5 in a 512 Hann window, 50% overlap. TOLs varied across frequencies and day hours. Light hours were noisier than dark hours in 25 frequency bands (MW, $p<0.01$), with highest mean level being measured at 794 Hz at 06am (105.4 $\pm$ 5.4 dB re 1μPa), and lowest mean level measured at 39.8 kHz at 01am (80.3 $\pm$ 3.7 dB re 1μPa). A total of 281 delphinid sound emissions were detected in nine occasions, seven during week 1 and two during week 2. Seven detection events occurred during dark hours. Although more than one species of delphinid is known to occur in Rio de Janeiro coast, these results indicate a frequent nocturnal use of the area, which was only possible to observe through autonomous monitoring.

9:15

4aAB6. Do nearby seismic airguns reduce singing effort in humpback whales? Michael J. Noad and Rebecca Dunlop (School of Veterinary Sci., The Univ. of Queensland, Gatton, QLD 4343, Australia, mnoad@uq.edu.au)

The high-level percussive sounds generated for seismic sea floor exploration have the potential to disrupt normal behaviors of whales. This study assesses if there is any reduction in the singing behaviour of migrating humpback whales in response to nearby airguns. Singing whales were acoustically tracked as they migrated along the coastline of south-eastern Queensland. A 20 cubic inch airgun or 140 cubic inch array of airguns was towed through the study area for 1 hour with the airguns firing every 11 sec (active treatments) or without the airguns operating (controls). Singing activity across each day was measured by counting the number of singing whales within the 10km-radius study area every 10 min, from 0700 to 1700, including during experiments. Singing activity during active periods and controls were compared with each other as well as with pseudo-randomly selected 1 hour periods when experiments were not underway (baseline). Changes in singing effort were also recorded by noting the number of instances of a singer stopping or starting during these periods. Preliminary analyses show no significant differences in singing activity, nor changes in singing effort, between active, control, or baseline periods.

9:30

4aAB7. Estimating received levels for acoustically tracked whales from Navy mid-frequency active sonar. Cameron R. Martin, Stephen W. Martin (National Marine Mammal Foundation, 2240 Shelter Island Dr. Ste. 200, San Diego, CA 92106, cameron.martin@nmmpfoundation.org), and E. E. Henderson (Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Multi-channel passive acoustic data were collected from bottom-mounted hydrophones off the coast of Kauai, HI at the U.S. Navy Pacific Missile Range Facility's instrumented range. In February 2017, data were recorded during the Submarine Command Course (SCC) training event that included mid-frequency active sonar (MFAS). Data were post-processed for minke whale (*Balaenoptera acutorostrata*) vocalizations and MFAS with automated detection, classification, and localization algorithms. Semi-automated processes were used to perform spatio-temporal associations of whale localizations to create individual whale tracks and to associate localized

MFAS transmissions with positional data from multiple ships. For each whale track, the received level from each MFAS transmission was estimated using a parabolic equation propagation model that statistically represented the uncertainty of whale location and depth. Exposures were accumulated over the duration of each whale track to estimate the cumulative sound exposure level (cSEL; dB re: $1\mu Pa^2 s$). Twenty-three minke whales were exposed to MFAS during the SCC; the maximum exposure was 169.8 dB cSEL, and the minimum distance to a ship transmitting MFAS was 4.8 km. Currently, statistical analyses of track kinematics and call characteristics are being conducted to quantify any significant changes between animal track kinematics with and without MFAS training presence.

9:45–10:00 Break

10:00

4aAB8. Effects of vessels and noise on the subsurface behavior of endangered killer whales (*Orcinus orca*). Marla M. Holt, Brad Hanson, Candice Emmons (NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112, Marla.Holt@noaa.gov), Jennifer B. Tennessen (Lynker Technologies, University Park, PA), Deborah Giles (Univ. of Washington Friday Harbor Labs, Friday Harbor, WA), and Jeffery Hogan (Cascadia Res. Collective, Olympia, WA)

Prey availability and disturbance from vessels and noise are identified threats to endangered Southern Resident killer whales. Vessel noise can mask echolocation signals used for hunting and/or disrupt foraging with implications for energy acquisition in a likely prey-limited population. We utilized suction cup-attached digital acoustic recording tags (DTAGs), to measure received noise levels, understand killer whales' use of sound, and determine effects of vessels/noise on subsurface behavior. During the 28 tag deployments, we collected vessel data concurrently along with opportunistic predation observations to validate feeding. Broadband received levels (dB re $1\mu Pa$) were significantly different across years. Of the vessel factors considered, both vessel count and speed were significant explanatory variables of received levels. Vessels emitting echosounder signals were commonly received by the DTAGs and overlapped with the echolocation frequencies use to hunt fish. Additionally, different phases of foraging were differentiated from the acoustic record, including the detection of crunching sounds after fish kills. Together with movement data analysis, these results allow the identification of different whale activities, including prey capture dives, to test hypotheses of vessel/noise effects on behavior. This work, along with a comparative investigation involving Northern Resident DTAG data, inform killer whale conservation and management measures.

10:15

4aAB9. Localizing bioacoustic signals with long-baseline hydrophone arrays. Benjamin T. Hendricks, T. Aaron Gulliver (Elec. and Comput. Eng., Univ. of Victoria, 3800 Finnerty Rd., Eng. Office Wing, Rm. 448, Victoria, BC V8P 5C2, Canada, hendrick@uvic.ca), Janie L. Wray (North Coast Cetacean Society, Hartley Bay, BC, Canada), Eric M. Keen (Marine Ecology and Telemetry Res., Seabeck, WA), Hussein M. Alidina (Oceans, World Wildlife Fund - Canada, Victoria, BC, Canada), Chris R. Picard (Gitga'at Oceans and Lands Dept., Hartley Bay, BC, Canada), and Hermann Meuter (none, Hartley Bay, BC, Canada)

Real-time localization and tracking of vocalizing marine mammals is a powerful tool to a) mitigate the risk of ship strikes and disturbance of the animals as well as b) to understand the impact of anthropogenic noise on habitat use of cetaceans. A long baseline hydrophone array has been installed in Squally Channel, a culturally, ecologically, and economically important marine environment in northern British Columbia, Canada. The array consists of four synchronized bottom-mounted hydrophones that permanently record and radio-transmit data to a land-based laboratory. Automated analysis tools have been developed to detect and localize transient bioacoustic signals from three or more hydrophones in real-time. They comprise the correlation of hydrophone signals, the construction of a replica surface using acoustic propagation models, and signal localization from a spatial likelihood surface. An overview of the localization method is presented. We assess spatial localization precision and discuss random and systematic sources of error stemming from the signal type and quality, bathymetry, sound speed profile, as well as the spatial distribution of hydrophones and acoustic sources. Simulation results are compared to field measurements. Based on our findings, challenges and opportunities associated with localization using long baseline arrays in coastal environments are discussed.

10:30

4aAB10. Multi-channel cross-correlation used to estimate time delays. Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 405, Honolulu, HI 96822, nosal@hawaii.edu)

Many marine mammal localization techniques rely on estimating the time of arrival delays of a call between the hydrophones of an array. This is commonly achieved by cross-correlating (some version of) received signals between hydrophone pairs. This process is complicated in cases with noise, multiple animals, and multipath arrivals. In this talk, I explore and demonstrate the potential of a multi-channel cross-correlation technique to help establish time delay estimates. Possible extensions to signal enhancement for detection and classification purposes are also discussed. [This work was supported by ONR Award No. N00014-16-1-2598.]

10:45

4aAB11. Turning birds into bats—Multi-modal tracking to study collective behaviour. Jens C. Koblitz, Oren Frokosh, Mate Nagy, Nora Carlson, Hemal Naik, and Couzin Iain (Max Planck Inst. for Ornithology, Univ. of Constance, Constance 78464, Germany, jkoblitz@orn.mpg.de)

Acoustic localization has been used to track numerous vocalizing animals, including whales, bats birds by measuring the time of arrival differences of a sound recorded by multiple receivers. This method, however, requires the species of interest to vocalize regularly to achieve decent temporal resolution. In order to track the movements of birds that do not vocalize regularly, a miniature, radio controlled ultrasonic speaker is attached to the birds and constantly emits ultrasound chirps. The chirps from various tagged animals can be discriminated based on information encoded in the signal. A 30 microphone array covering the ceiling of a large aviary ($14.8 \times 6.6 \times 3.9$ m) is used to record the chirps and provides the basis for accurate localization of the sources. As the chirps do not overlap with the frequency of the animals vocalizations, these can be localized and assigned individually in a flock of moving birds. In addition, a VICON motion capture system provides a simple and accurate method to ground truth the acoustic localizations as it records very precise movement information of the individuals while line of sight is between the marker on the animal and a number of cameras is established. The acoustic and visual tracking systems working together provides a unique and novel answer to consistent and precise localizations of non-vocalizing individuals (and groups of individuals) as they move through space.

11:00

4aAB12. Increasing access to big bioacoustic data through cloud-based systems. Carrie C. Wall, Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), Allyson Spring (Cloud Platform, Google, Alexandria, VA), and Jason Gedamke (Sci. & Technol., NOAA NMFS, Silver Spring, MD)

The National Oceanic and Atmospheric Administration's (NOAA) National Centers for Environmental Information (NCEI) has recently developed an archive for the long-term stewardship of passive acoustic data. Protocols for archiving passive acoustic data are currently being established in support of the NOAA Ocean Noise Reference Station Network project, and monitoring marine mammals and fish. Archives maintain data, but access to these data is a core mission of NCEI that allows users to discover, query, and analyze the data in new and innovative ways. To facilitate global access to what will be 100s of TB of bioacoustic data, NCEI has partnered with Google Cloud through the NOAA Big Data Project. Cloud-based access to large volumes of data creates the ability to listen to sound files, download more easily, and bring processing routines to the data instead of bringing the data to the processing.

11:15

4aAB13. Acoustic signature and footprint of artificial nurseries in harbors. Julie Lossent (Res. Institut CHORUS, 22 rue du Pont Noir, Saint Egrève 38120, France, julie.lossent@chorusacoustics.com), Anais Gudefin (Ecocean, Montpellier, France), Cédric Gervaise (Res. Institut CHORUS, Grenoble, France), Amélie Foncuberta, Lecaillon Gilles (Ecocean, Montpellier, France), and Lucia D. Iorio (Res. Institut CHORUS, Grenoble, France)

Underwater soundscapes are used as a cue by vertebrate and invertebrate larvae and post-larvae to locate suitable settlement habitats. In the project SEANAPS, we investigated if soundscapes can play a role in ecological restoration actions. We assessed whether Biohuts® (artificial nurseries installed in harbors) have a distinct acoustic signature from the harbor and the natural environments. We conducted four acoustic measurement sessions in Cap d'Agde (France) and a simultaneous monitoring of vagile fauna present in Biohuts®. Directional listening networks allow us to locate the broadband transient sounds from benthic invertebrates and fish, build maps of these emissions and isolate the biophony emitted by the vagile fauna of Biohuts®. The Biohuts® produced 78 emissions/min/m^2, which represent 200 and 130 times more than a pseudo-natural habitat (seawall) and a natural rocky habitat, respectively. The acoustic footprint of the Biohuts® signatures is 120 m radius in the harbor and 500 m in the harbor road. The harbor road is a naturally silent corridor, which could slow down the larval recruitment by its lack of attractive acoustic signature. This information can be used to resize the network of Biohuts®. Moreover, we demonstrated that the acoustic signature of a Biohut® informs about the vagile fauna it hosts.

11:30

4aAB14. Riverscape meets underwater soundscape: Acoustic habitat selection by brook char in a small stream. Zaccaria Kacem, Marco A. Rodríguez, and Raphaël Proulx (Environ. Sci., Univ. of Quebec in Trois-Rivières, 575 rue des forges, app2, Trois-Rivières, QC G9A 5W4, Canada, zaccaria.kacem@uqtr.ca)

Stream habitats are characterised by geophysical descriptors such as water temperature, depth, substrate type, and flow speed. So far, few studies have focused on underwater sounds as an important feature of habitat selection by fish. In this study, we described stream habitats at high resolution to evaluate the relative importance of the underwater soundscape and other geophysical descriptors for understanding the distribution of brook char densities. Our results showed the high acoustical heterogeneity of stream habitats (ranging from 40 dB up to 150 dB re 1uPa), which was related to differences in water velocity and depth as expected from theory. Brooks char densities were nevertheless positively related to sound intensities, irrespective of water velocity, depth, or facies type. Our findings showed that underwater sounds integrate the many environmental dimensions of stream and may be used as cues for habitat selection. The positive relationship between brook char densities and sound intensities could be related to the high auditory threshold of Salmonidae.

11:45

4aAB15. Long-term characterization of the marine soundscape in Kona, Hawaii. Karlina Merkens (CRP, NOAA/PIFCS (Lynker Tech.), 3710 SW Caldew St., Portland, OR 97219, kmerkens@gmail.com), Erin M. Oleson (NOAA/NMFS/PIFSC, Honolulu, HI), and Simone Baumann-Pickering (UCSD/SIO, La Jolla, CA)

Soundscape analysis of marine environments is still a new field of inquiry, with the majority of studies focusing on individual species or acoustic functional groups. Additionally, the challenges of making long term acoustic recordings in marine habitats, particularly in deep water, have limited the duration of many analyses. The Pacific Islands Fishery Science Center has been collecting passive acoustic data at a depth of 630 m off the Kona coast of the Island of Hawaii since 2007, and has recently begun a soundscape analysis effort to characterize the contributors to the local acoustic environment. This project seeks to examine not only low frequency noise patterns, below 1 kHz, but also higher frequency sounds that may overlap in frequency with the signals of many cetacean species. We have begun by identifying the relative contribution of odontocete echolocation, and also close approaches of boats and the presence of echosounders. The temporal patterns in these signals reveal daily and seasonal cycles in biological and anthropogenic sounds, which provide both detailed insight into the interactions between humans, animals and their environment, and long term trends in activity of those sound sources.

THURSDAY MORNING, 8 NOVEMBER 2018 SIDNEY (VCC), 9:15 A.M. TO 12:00 NOON

Session 4aBA

Biomedical Acoustics: Biomedical Acoustics I

Tyrone M. Porter, Chair
Boston University, 110 Cummington Mall, Boston, MA 02215

Contributed Papers

9:15

4aBA1. Preclinical assessment and enhancement of passive acoustic mapping in clinically realistic propagation environments. Michael Gray and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX37DQ, United Kingdom, michael.gray@eng.ox.ac.uk)

Passive acoustic mapping (PAM) techniques have been developed to detect, localize, and quantify cavitation activity during therapeutic ultrasound treatments. Building on a decade of *in vitro* and small animal studies, this paper presents a series of pre-clinical experiments and simulations conducted in preparation for an upcoming clinical trial of ultrasound-mediated targeted drug delivery. The effects of tissue attenuation and refraction, array shape and element diffraction, soundspeed uncertainty, and imaging algorithm will be discussed. Critically, techniques will presented for removing factors that would otherwise bias PAM results both within and between patients. Together, the results indicate the potential to significantly enhance both the qualitative and quantitative capabilities of PAM for ensuring clinical therapeutic safety and efficacy. [Work supported by the National Institute for Health Research Oxford Biomedical Research Centre.]

9:30

4aBA2. Attenuation compensation of ultrasound backscatter from human carotid plaque. Sheronica L. James, Russell Fedewa (Biomedical Eng., Cleveland Clinic, 9500 Euclid Ave., ND-20, Cleveland, OH 44195, jamess4@ccf.org), Heather Gornik (Cardiovascular Medicine, Cleveland Clinic, Cleveland, OH), Sean Lyden (Vascular Surgery, Cleveland Clinic, Cleveland, OH), and D. Geoffrey Vince (Biomedical Eng., Cleveland Clinic, Cleveland, OH)

Carotid atherosclerotic plaque composition may be a valuable predictor of stroke risk. Here, a patient adaptive attenuation compensation technique is introduced. A Siemens S3000, 9L4 probe and Axius Direct software were used to acquire radiofrequency (RF) data from 19 subjects prior to carotid endarterectomy. Histology slices of the excised plaque were prepared and matched to ultrasound frames. Regions of interest (ROI) were selected from a homogeneous area within the slide stack and matched to corresponding ROI's in the RF data. ROI's were categorized as fibrous (n = 32), hemorrhagic and/or necrotic core (n = 74), or calcium (n = 32). Additionally, 209 adventitia ROI's were obtained from six normal subjects. ROI power spectra were computed and normalized to a uniform phantom. Five attenuation compensation methods were applied to the spectra: (1) reference phantom with 0.5 dB/cm-MHz attenuation; (2) nominal power spectral shift estimator; (3) optimum power spectral shift estimator; (4) normalized backscatter from adventitia; and (5) patient adaptive two-step attenuation compensation. A linear fit of the resulting estimated backscatter transfer functions (*eBTF*) was performed over the fundamental bandwidth of 3–7 MHz. Only the patient adaptive two-step attenuation compensation distinguished among the means of all four tissue types for the linear mid-band fit parameter.

9:45

4aBA3. Beam coding with orthogonal complementary Golay codes for signal-to-noise ratio improvement in ultrasound mammography. Yasin Kumru and Hayrettin KÖymen (Elec. and Electronics Eng., Bilkent Univ., Ankara 06800, Turkey, yasin.kumru@bilkent.edu.tr)

In this paper, the experimental results of using Golay coded signals at 7.5 MHz to detect breast microcalcifications of 50 μm size is presented. For improving the success on detecting the microcalcifications, orthogonal Golay complementary sequences up to 40 chips having cross correlation for minimum interference are used as coded signals and they are compared to tone burst pulse of equal energy in terms of resolution under weak signal conditions. The measurements are conducted using an experimental ultrasound research scanner, DiPhAS having 256 channels, a phased array transducer with 7.5 MHz center frequency, and the results obtained through experiments are validated by Field-II simulation software. In addition, to investigate the superiority of coded signals in terms of resolution, multipurpose tissue equivalent phantom (Model 84-317) containing series of monofilament nylon targets, 240 μm in diameter, and cyst-like objects with attenuation of 0.5 dB/(MHz*cm) is used in the experiments. We obtained ultrasound images of monofilament nylon targets and cyst-like objects for the evaluation of resolution. Experimental results show that it is possible to differentiate closely positioned small targets with increased success by using coded excitation in very weak signal conditions.

10:00

4aBA4. Experimental reconstruction of sound speed and attenuation profiles. Pedram Mojabi and Joe LoVetri (Elec. and Comput. Eng., Univ. of MB, 75 Chancellor's Circle, Winnipeg, MB R3T5V6, Canada, pedram.mojabi@gmail.com)

Quantitative reconstruction of sound speed and attenuation profiles using ray-based approaches are considered. To this end, experimental datasets from two different ultrasound imaging systems are utilized. The first is our in-house system that is composed of a relatively small number of co-resident fixed piezo-electric transducers. The second, referred to as the MUBI system [Camacho, Archives of Acoustics, 2012], is equipped with many more transducers that can be rotated around the object-of-interest, allowing higher resolution reconstructions. The raw data collected from our system cannot be directly used in the reconstruction algorithms because various sources of noise and systematic errors corrupt the data. We compare the image reconstruction performance when different preprocessing techniques are applied to the raw data, as well as when various techniques of determining the time-of-arrival, such as the Akaike information criterion (AIC), are employed. The quantitative ray-based reconstruction algorithms are tested on simplistic tissue-mimicking phantoms [Mojabi & LoVetri, Submitted to IEEE UFFC, 2018].

10:15–10:30 Break

10:30

4aBA5. Estimation of the thermo-physical property of biological tissues from temperature rise due to ultrasound exposure. Yukako Tsujimoto and Iwaki Akiyama (Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, ctub1049@mail4.doshisha.ac.jp)

This study discusses the feasibility of a method for ultrasonic tissue characterization according to the thermo-physical properties of biological tissues. Since this method measures the ratio of sound velocity variation due to ultrasonic heating, it is significant to clarify the relationship between the ratio and thermo-physical properties. The ratio of sound velocity variation of a tissue sample was measured from phase shifts of echoes before and after the temperature rise by transmitting an ultrasonic pulsed wave. In the experiments the transducer used for heating was a resonance frequency of 5.0 MHz, concave ring shape, inner diameter of 5 mm, and outer diameter of 12 mm. It was arranged coaxially on the outer perimeter of the transducer used for the measurement. This transducer was a resonance frequency of 10 MHz, a focal distance of 15 mm, and a diameter of 4 mm. The exposure time of heating ultrasound is 100 ms. As a result of heating sound pressure of 1.0 MPa, 1.5 MPa and 2.0 MPa, the estimated values of temperature rise were 0.56 °C, 1.1 °C and 2.0 °C, respectively. [This study was supported by MEXT-Supported Program for the Strategic Research Foundation at Private Universities, 2013-2017.]

10:45

4aBA6. Experimental and numerical investigation of backscattered signal strength from different concentrations of nanobubble and microbubble clusters. Hossein Haghi, Amin Jafari Sojahrood (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2k3, Canada, hossein.haghi@ryerson.ca), Al C. De Leon, Agata Exner (Biomedical Eng., Case Western Reserve Univ., Cleveland, OH), and Michael C. Kolios (Phys., Ryerson Univ., Toronto, ON, Canada)

Optimizing the clinical performance of Nanobubbles and Microbubbles (MBs and NBs) requires not only a good understanding of their individual bubble dynamics, but how bubble interactions change these dynamics. Here we report our experimental and simulation results showing the changes in backscattered signal strength from different concentrations of NBs or MBs. NBs and MBs are sorted by size to isolate two populations with mean diameters of 200 nm and 1 micron, respectively. Our results show that increasing number density of NBs or MBs is accompanied with an increase in the strength of their backscattered signals up to a certain concentration, above which the strength of the signal decreases. This decrease is not associated with the medium attenuation which was not accounted for. Our numerical simulations that include inter-bubble interactions are in general agreement with our experimental results. Moreover, acquired data show the that concentration at which there is signal saturation for NBs is orders of magnitude higher than that of MBs (1e9 vs 1e6 bubbles per mL), suggesting a potential explanation behind the very strong backscatter observed in NB clusters compared to MBs. Numerical simulations show that increasing the number density of bubbles decreases the individual signal strength of each individual bubbles.

11:00

4aBA7. Characterization of perfluorocarbon nanodrops with 18-MHz plane waves. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10006, jkettermeister@gmail.com), Tiffany-Trang Nguyen, and Mario L. Fabiilli (Dept. of Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Perfluorocarbon (PF) nanodrops (NDs) undergo a phase change from liquid to vapor with an acoustic stimulation. Activation thresholds are established with relatively high-concentrations of NDs. Activating single NDs with high-frequency plane waves without the need to switch between transmit waveforms would allow uninterrupted high-speed imaging studies to be undertaken in microvasculature. Lipid-coated perfluoropentane (PFP) or perfluorohexane (PFH) NDs were made via sonication (S) or sonication and extrusion (SE). A control sample consisted of lipid blend (LB) without PF. A Verasonics Vantage with an 18-MHz array was used to acquire backscatter data from activated NDs. A test tank with a 200-μm diameter input flow channel was used to pass a diluted concentration of NDs into the image plane of the array. The injection flow rate was 55 mL/hr. Batches of 5 plane waves were transmitted over $\pm$ 5 degrees at an effective PRF of 1000 Hz. Data were collected over 10 s intervals and then processed to count activation events. An event was defined as a backscatter signal 25 dB above background noise. PFPSE showed the highest counts at a given pressure. Agents appeared to be activated prior to entering the image plane. PFPS cases showed a reduced rate of activation but with a similar linear rise. PFH and LB had minimal to no events. The data indicate a small fraction of the injected agents were activated but do not indicate a distinct activation threshold was reached.

11:15

4aBA8. Influence of surfactant encapsulation on the acoustic droplet vaporization. Thomas Lacour, Tony Valier-Brasier, and François Coulouvrat (Institut Jean Le Rond d'Alembert, Sorbonne Université CNRS UMR 7190, 4 Pl. Jussieu, Paris 75005, France, thomas.lacour@upmc.fr)

Nanodroplets have great, promising medical applications such as contrast imaging, embolotherapy or targeted drug delivery. Their functions can be mechanically activated by means of focused ultrasound inducing a phase change of the inner liquid known as acoustic droplet vaporization (ADV) process. In this context, a four-phases (vapor + liquid + surfactant layer + surrounding environment) model of ADV is proposed. Attention is especially devoted to the nonlinear mechanical properties of the surfactant layer encapsulating the metastable bubble/droplet system and described as a model of nonlinear surface tension with two parameters. Various responses to ultrasound excitation are illustrated, depending on surfactant properties and acoustical excitation parameters. Different classes of ADV outcomes are exhibited, and a relevant threshold ensuring complete vaporization is defined. The dependence of this threshold with acoustical, geometrical, and mechanical parameters is also discussed.

11:30

4aBA9. Phase shift nanoemulsions facilitated focused ultrasound non-thermal ablation in mice brain. Chenguang Peng (Biomedical Eng., Boston Univ., Boston, MA), Tao Sun, Natalia Vykhodtseva, Yongzhi Zhang, Chanikarn Power, Nathan McDannold (Radiology, Brigham Women Hospital, Boston, MA), and Tyrone M. Porter (Biomedical Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

High intensity focused ultrasound (HIFU) thermal ablation is an emerging technique for noninvasive transcranial surgery. One of the major issues of transcranial HIFU thermal ablation is that the pressure and power required to generate lesions is relatively high and could induce skull heating and pain in some cases. Artificial nuclei like phase shift nanoemulsions (PSNE) have been shown to significantly lower the acoustic pressure and power needed to create lesions through inertial cavitation. However, traditional PSNE made from dodecaperfluoropentane (DDFP) were not able to activate below a peak negative pressure (PNP) of 3.5 MPa, which may be difficult to achieve inside the brain. In this study, we proposed to use a more volatile perfluorocarbon—perfluorobutane (PFB)—to synthesize PSNE so that they could be activated at much lower pressures. A transcranial focused ultrasound transducer operating at 740 kHz and 12 CD-1 mice were used to test the vaporization of PFB PSNE and the corresponding damage. The PNP was increased stepwise up to 1.8 MPa and the broadband emission from the inertial cavitation nucleated by vaporized PFB-based PSNE were recorded to quantify the inertial cavitation level. A significant elevation of broadband emission was noticed first at PNP = 1.25 MPa and increased dramatically, indicating strong inertial cavitation. The corresponding biological damage was analyzed with H&E staining and exhibited confined ischemic and hemorrhagic lesions.

11:45

4aBA10. Novel sonification and haptic displays of physiologic variables to improve patient safety. Joseph Schlesinger, Taylor Combs, Megan Holmberg, Seiver Jorgensen, and Samantha Kultgen (Vanderbilt Univ. Medical Ctr., 1211 21st Ave. South, Medical Arts Bldg., #526, Nashville, TN 37212, joseph.j.schlesinger@vanderbilt.edu)

Auditory medical alarms in healthcare are uninformative, poorly localizable, have a low positive predictive value, and are set at thresholds forcing practitioners to be reactive instead of proactive. Sonifying patient physiology as it transitions from normal to abnormal will allow practitioners to respond before the patient status devolves into an emergency. Multimodal interactions have the potential to enhance future sonification tools by reducing alarm fatigue—the addition of haptic cuing to novel physiologic sonification may improve patient safety. To increase the overall efficiency of alarm design in healthcare, we propose reducing alarm fatigue by integrating multisensory streams. We present an experiment that characterizes how the integration of tactile and auditory signals affects the speed and accuracy of alarm responses. Participants received multisensory auditory and haptic input while performing a task designed to tax attentional resources mimicking working in the ICU. Our results indicate there is a trend towards increased perception of change of physiologic variables with concordant haptic stimuli and participants are significantly better at determining the direction of change versus the physiologic variable of change. Future directions include simplification of the sonification schemata and increasing information complexity in the haptic modality to utilize multisensory integration.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics

Eric Rokni, Chair
Department of Physics, Rollins College, Winter Park, FL 32789

Contributed Papers

8:55

4aMU1. Are phantom partials produced in piano strings? Thomas Moore, Lauren Neldner, and Eric Rokni (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The anomalous frequency components in the sound from a piano, often referred to as phantom partials, were first noted almost 75 years ago. They occur at the sums and differences of the frequencies generated by the transverse motion of the string, and it is generally accepted that they are produced by longitudinal motion of the string that is associated with the normal transverse motion. Experimental evidence will be presented demonstrating that while some of the power in a phantom partial may originate in the string, most of it does not. A simple theory explaining how phantom partials may originate in the wooden components of the piano will be presented.

9:10

4aMU2. Overtone series in variable density strings. Greg S. Elliott (Phys., Univ. of Puget Sound, 1500 N Warner CMB 1031, Tacoma, WA 98416, gelliott@ups.edu), Martin Jackson, and Ethan Russell (Mathematics and Comput. Sci., Univ. of Puget Soud, Tacoma, WA)

The vibrational mode frequencies of a taut string depend on the form of its mass distribution. To lowest order, deviations away from a uniform mass density that are even about the center of the string cause shifts in the overtones away from harmonic, while deviations that are odd do not. The problem of determining the overtone series for a given mass distribution is formulated and a full solution is derived for variations in the inverse mass density that are expressed as a Fourier cosine series. Variations arising from a single Fourier component predominantly shift the relative frequency of a single mode, with the $2n^{th}$ Fourier component shifting the frequency of the n^{th} mode. For a physically reasonable range of mass density variations, the frequency of a mode can typically be shifted by two semitones. Wrapped metal strings with variations in inverse mass density with different Fourier components have been fabricated, characterized, and tested. Light plucking produces overtones in agreement with the model; ordinary plucking produces incommensurate tones at sum and difference frequencies. Overtone series with several different major and minor tones are possible for variations involving more than one Fourier component.

9:25

4aMU3. How different strings affect violin qualities. Lei Fu, Gary Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, lei.fu2@mail.mcgill.ca), and Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Unité Mixte de Recherche 7190, Sorbonne Université / Ctr. National de la Recherche Scientifique, Paris, France)

The brand and model of strings used on violins are considered to play a significant role in their playability and sound quality. An experiment was designed to test the perceptual quality of different violin strings. A professional violinist selected two violins, from a set of the same make/model, that had similar sound and playing qualities. Three different types of strings were chosen for this study: Dominant, Kaplan, and Pro-Arté strings. Professional and advanced student violinists were invited to play and evaluate the violins. The experiment involved three phases: in the first phase, the two violins were strung with the same types of strings; in the last two phases, the strings of one of the violins was changed to the other two different types of strings (in a random manner). Violinists were asked to freely describe the differences of the two violins in each phase and rate them on eight specified criteria. Preliminary results indicate that the violin with Dominant strings was perceived as being more responsive and having a brighter sound, while that with the Pro-Arté strings was considered to have a richer sound and better overall quality.

9:40

4aMU4. Perceptual thresholds of violin soundpost length variation. Lei Fu (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, lei.fu2@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Unité Mixte de Recherche 7190, Sorbonne Université / Ctr. National de la Recherche Scientifique, Paris, France), and Gary Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

The soundpost of a violin is an essential component of the instrument, providing structural support and a mechanical coupling between top and back plates. Subtle changes made by a luthier to the soundpost dimensions or position can result in significant variations in sound and playing qualities. A study was designed to investigate the perception of sound quality associated with changes to the soundpost length, making use of an adjustable carbon fiber soundpost. Violin excerpts were recorded with different soundpost lengths. Violinists and luthiers were invited to listen to the recordings and provide perceptual feedback using a computer interface. Tasks included pairwise comparisons and free sorting. Perceptual thresholds in detecting soundpost length changes were characterized, as well as possible groupings according to soundpost lengths that might be considered too short, too long and optimal. Bridge admittances were also measured and analyzed for different soundpost lengths.

9:55

4aMU5. Spherical harmonic expansions of high-resolution directivity data. Samuel D. Bellows and Timothy W. Leishman (Brigham Young Univ., 910 N 900 E Apt. 111, Provo, UT 84604, sbellows@byu.edu)

Quantifying the directivity patterns of sound sources provides not only deeper insights into understanding the source and its acoustical properties but also the potential to simulate the propagation of its sound in various settings. Being able to describe the detailed 3-dimensional directivity of highly complex and dynamic sound sources such as musical instruments and human speech in a simple way while at the same time preserving the fine details of the directivity patterns has broad applications. By using seventeenth-order spherical harmonic expansions of data collected on a 2,664-

point spherical array, large amounts of the measured directivity of musical instruments and human speech was condensed in a simple, accurate, and powerful tool for understanding these complex systems.

10:10

4aMU6. Studying the clarinet with an artificial mouth: Comparison of playing frequencies between model and measurement. Jack D. Gabriel and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, jgabriel@rollins.edu)

In the field of musical acoustics, instruments, such as the clarinet, are often played with the use of an artificial mouth or playing machine in order to objectively measure the playing characteristics such as sound levels, playing frequency, regime changes, etc. The purpose of this research was to study the tuning tendencies of the clarinet experimentally and compare this to the models found in literature. A clarinet was artificially blown to determine the playing frequencies for varying levels of blowing pressure and reed opening. These measurements could then be compared to computational models of the clarinet. The experimental results will be presented and compared with the theoretical values from past studies that predicted playing frequencies both analytically and computationally. Finally, suggestions for improving the models will be presented and discussed.

10:25–10:45 Break

10:45

4aMU7. Construction and preliminary studies of the 3-6-(7) tones Oja. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu, UEnugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

Common in the eastern parts of Nigeria among Igbos, the oja is a Nigerian wooden flute. More like a whistle with its rich sharp tone, the oja is an end blown flute about 14–16 cm long with a narrowed, hollowed cavity running down its entire length and another across its width. These form three holes that can be covered in various combinations with the fingers of both hands, giving the flute a range of about a sixth or a seventh depending on the master player. The mouthpiece is "U" or "V" shaped. With the three holes, it is expected that there would be three distinct pitches produced from the "oja" but more than three tones can be heard or sounded. This paper showcases the construction of oja flute, and presents preliminary studies on how the sixth can be realized. Pictorials, data, and videos of practical experience are presented to support the theoretical discourse.

11:00

4aMU8. Construction and validation of a high resolution finite element model of the Japanese *koto*. Angela K. Coaldrake (Elder Conservatorium of Music, Univ. of Adelaide, Adelaide, SA 5005, Australia, kimi.coaldrake@adelaide.edu.au)

The Japanese *koto* is recognized for its complex resonances, but few studies have been conducted to understand its acoustic properties. This paper presents a COMSOL Multiphysics finite element model of a handcrafted, professional-grade Japanese *koto*. To do this, the complex internal and external geometry was captured on 2400 CAT scan cross-sections and imported into Comsol. The model was validated by reference to literature data, Chladni patterns, frequency response experiments, acoustic camera and laser scanning vibrometer studies. These results were then compared to the sound as played on the actual instrument. Challenges in the model's development arising from the materials and construction of the instrument are discussed, most notably the anisotropic nature of the *paulownia* wood including multiple grain orientations used in its construction and the added complexity of modeling the organic shape of the real, hand-crafted 1.83 m long instrument with its major internal variations. These developments were guided by parallel studies in less intractable geometry such as simpler box models or models with idealized geometrical shapes lofted along a spline. The CAT scan derived model is used as a quasi-experimental tool to investigate the effect of internal components of the *koto* on its resonances and results presented.

11:15

4aMU9. Kuramoto and Sommerfeld: Toward comprehension of symphony orchestras in concert halls. James B. Lee (None, 6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Given comprehension of physical acoustics in concert halls, an accurate replacement to the inadequate concatenation of reverberation theory, geometric acoustics, and Beranek's "perceptual categories" that long has held the field in architectural acoustics, it remains to explain how the symphony orchestra couples to the space. Individual instruments do not couple to the space directly, but first to others in their respective sections: first violins to first violins, second violins to second violins, violas to violas, cellos to cellos, basses to basses, and so on through the winds, and others. Each section is synchronized in phase by a Kuramoto mean-field. Then the coherently radiating sections couple to each other asynchronously and create a unique propagating field according to the Sommerfield radiation condition. Kuramoto mean-field coupling is key, for absent that a section's individual instruments would tend to randomize in phase and sum to zero power. The Sommerfeld radiation condition does not guarantee such a uniform field, only a unique one, the power and information of which are conveyed to the audience as constrained by the usual Gabor-Huygens principle: proximity effects and resonant scattering are essential.

11:30

4aMU10. Human-technology interfaces with the tactile metronome. Nathan Hesselink (Univ. of British Columbia, UBC School of Music, 6361 Memorial Dr., Vancouver, BC V6T 1Z2, Canada, n.hesselink@ubc.ca)

The story of the metronome is the story of humankind coming to terms with evolving conceptions of time and coordination as mediated through technologies of the modern age. What began as a tool for regulating and documenting tempo soon became the temporal yardstick by which aspiring musicians strove to emulate in practice and performance. In the early twentieth century the metronome took on the new role of synchronizing live musicians with moving images on a screen (the so-called "click track"), and as the century progressed the metronome would come to dictate the manner in which musicians related to each other in the recording studio and in live events. This paper focuses on the latest manifestation of this phenomenon, the tactile metronome, looking at how vibrotactile stimulation is being used for temporal synchronization as well as an enhanced sense of embodiment for the performer, including haptic feedback. Four modern products tailored to performing musicians will be introduced and analyzed in the overlapping contexts of synchronization and embodiment. It is my argument that the metronome has now come full circle, returning a sense of human feel to the experience of making music.

Session 4aNS

Noise, Speech Communication, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance I

Joonhee Lee, Cochair
Department of Building, Civil and Environmental Engineering, Concordia University, EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada

Z. Ellen Peng, Cochair
Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. Energetic and informational masking effects of hospital noise. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

The ability to understand speech in a hospital setting is limited by the overlap of speech and noise energy at the level of the cochlea, sometimes called "energetic masking." However, there are also additional "informational masking" effects that can occur when the noise is comprised of informational sounds such as speech, alarms, or other meaningful auditory events (e.g., Kidd and Colburn, 2017). While energetic overlap has been well-studied and is quantified in a number of international standards such as the Speech Intelligibility Index (ANSI, 2004), there is no similar metric for informational masking. Two of the main difficulties with predicting informational masking, and thus the intelligibility of speech in real world environments, are 1) the lack of tools that defines the meaningfulness of an interfering sound and 2) the much greater variability in susceptibility of individual listeners to informational as opposed to energetic masking. These ideas will be illustrated with data from several recent studies of speech intelligibility with various types of maskers, including hospital noise, and several current and emerging techniques for assessing speech intelligibility for individual listeners in simulated environments will be described.

8:25

4aNS2. Do orienting responses predict annoyance from irrelevant sounds? Jordan N. Oliver (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN), Lea Sollmann, Annika Niehl (Systems Neurosci. & NeuroTechnol. Unit, Univ. of Appl. Sci., Saarbruecken, Germany), and Alexander L. Francis (Speech, Lang. and Hearing Sci., Purdue Univ., Speech, Lang. and Hearing Sci., West Lafayette, IN 47907, francisa@purdue.edu)

Unexpected sounds are distracting and can be annoying but individuals may differ in susceptibility to them. Irrelevant sounds occurring at sparse temporal intervals induce a psychophysiological orienting response reflecting involuntary capture of attention away from the primary task. We hypothesize that the frequency and/or magnitude of individual listeners' orienting responses to irrelevant sounds will predict annoyance ratings and task performance in distracting noise. Participants read essays while seated in a comfortable chair in a sound-shielded booth facing a semicircular array of 6 speakers located 1.5 m away at 30°, 60°, and 90° to the left and right. Unintelligible background speech (ISTS) played at 60 dB(A) SPL from each loudspeaker (unsynchronized). At 50–70 s intervals one of 12 non-speech sounds (IADS) played for 6 s from one loudspeaker at approximately 70 dB(A) SPL. Order and location of sounds were randomized but each sound played from each speaker exactly once over the experiment (72 trials, ~80 min). Cardiovascular, electrodermal, electro-ocular, and bilateral posterior auricular muscle activity were recorded from participants to quantify orienting response. Behavioral measures of reading comprehension, noise sensitivity, personality traits, and subjective effort, frustration and annoyance were also collected and will be related to physiological measures.

8:45

4aNS3. Spatial release from masking in reverberation for children and adults with normal hearing. Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, z.ellen.peng@wisc.edu), Florian Pausch, and Janina Fels (Medical Acoust. Group, Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, NRW, Germany)

The ability to perform spatial release from masking (SRM), i.e., benefiting from a spatial separation between target and distracting talkers, is an important developmental skill for children to navigate noisy environments such as classrooms. While studies on children's SRM were mostly conducted in anechoic rooms or sound booths to date, little is yet known how realistic room acoustics including

reverberation in everyday listening affect this ability. In this study, we measured SRM from children and adults with normal hearing in virtual acoustic environments that mimic typical classrooms with different acoustic conditions. Two virtual classrooms were simulated with mean mid-frequency reverberation times (RTs) of 0.4 and 1.1 s, one within and the other poorer than the recommendation of classroom acoustics standard (ANSI S12.60). Overall, children performed more poorly than adults on both speech intelligibility and SRM. Children's speech intelligibility of the target speech masked by distractor voices was better under the less reverberant condition. Interestingly, while adults showed better SRM under RT = 0.4 s, children did not perform any differently between the two RT conditions. The findings will be discussed for further investigation and implications for classroom acoustic designs. [Work supported by the EU 7th Framework Programme, ITN FP7-607139.]

9:05

4aNS4. Children's real-time spoken language processing in the presence of background speech. Tina M. Grieco-Calub, Katherine M. Simeon, and Shana Birger (Northwestern Univ., 2240 Campus Dr., FS, 2-231, Evanston, IL 60208, tinagc@northwestern.edu)

Children's naturalistic environments often contain noise, including background speech and environmental sounds, that disrupts their speed and accuracy of spoken language processing. Integrating congruent auditory and visual speech cues, a skill that improves throughout childhood, facilitates language processing in background noise. The present study implemented an integrated eye-tracking and touch-screen paradigm to explore how children utilize visual speech cues in the presence of background speech and how these behaviors change with age. Typically-developing children (ages 3–12 years), either heard (auditory-only) or heard and viewed (audiovisual) a female speaking sentences (e.g., *Find the dog*) in quiet or in the presence of a male two-talker speech masker at + 2 dB SPL. Children were then instructed to select an image, among a set of three, that matched the sentence-final word. During each trial, children's eye gaze was recorded, which allowed us to quantify children's fixations to the target versus the distractor images. These data enable us to make fine-grained measurements of children's language processing in real time. Performance was also quantified by accuracy of target image selection. Discussion will focus on children's spoken language processing in the presence of background speech and in the presence and absence of congruent audiovisual speech cues.

9:25

4aNS5. Effects of background noise in vowel productions of children with cochlear implants. Areti Okalidou (Educational and Social Policy, Univ. of Macedonia, 156 Egnatia St., P.O. Box 1591, Salonika 540 06, Greece, okalidou@uom.edu.gr), Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin, Madison, WI), Polina Pantazidou (Educational and Social Policy, Univ. of Macedonia, Salonika, Greece), Janina Fels (Inst. for Tech. Acoust., RWTH, Aachen, Germany), Michalis Nistikakis (Educational and Social Policy, Univ. of Macedonia, Salonika, Greece), and George Kyriafinis (Dept. of Medicine, 1st University ENT Clinic of Ahepa Hospital, Aristotle Univ. of Thessaloniki, Salonika, Greece)

Early cochlear implantation led to significant improvements in the speech of deaf children, yet, reduced accuracy and greater token-to-token variability are noted. Vowel spaces are either normal or reduced and vowels are produced with less consistency (e.g., Baudonck *et al.* 2011; Neumeyer *et al.* 2010). So far, studies investigated speech in quiet. In the real world, children with cochlear implants (CI) function in noisy environments with auditory feedback containing speech and noise which is known to alter speech production even in typically developing adults (e.g., Bottalico *et al.*, 2015). Since children with CI produce less robust vowels, this study aimed to examine the effects of everyday noise on their productions. Eight children with CI, 6;10–12;0 years old, produced words with Greek vowels while listening to auditory stimuli provided directly via auxiliary input port to the speech processor. The conditions were: a) quiet, b) speech-shaped noise, and c) speech-shaped noise with reverberation. Acoustic measurements of F1 and F2, pitch, duration and intensity were made. Apart from inter-subject variability, results indicated changes in pitch, F1, and intensity with added noise along with occasional changes in duration and F2. Individual analysis focused on age and vowel space changes as the acoustic conditions changed.

9:45

4aNS6. Relationship between perception of speech in noise and disordered speech: Influence of sensorineural hearing loss. Sarah E. Yoho and Stephanie A. Borrie (Communicative Disord. and Deaf Education, Utah State Univ., 1000 Old Main Hill, Logan, UT 84321-6746, sarah.leopold@usu.edu)

It is well known that there exists substantial individual variability in the ability to understand speech in adverse listening conditions. Despite this variability, a recent study with normal-hearing listeners has revealed a strong relationship between the ability to perceive speech in noise (environmental degradation) and dysarthric speech (source degradation) [Borrie *et al.*, *Journal of Acoustical Society of America,*141, 4660–4667 (2017)]. While a large body of literature on the difficulty faced by hearing-impaired listeners in understanding speech in noise exists, the difficulties faced by this population in understanding dysarthric speech has received much less attention. Further, investigations into the relationship between processing speech in noise and dysarthric speech for listeners with hearing loss do not exist. This current study extends on previous findings, investigating the relationship between processing speech in noise and dysarthric speech for listeners with sensorineural hearing loss. Preliminary results replicate previous findings of a relationship between the ability to perceive speech in noise and dysarthric speech for normal hearing listeners. This relationship is also observed for hearing-impaired listeners; however, these listeners perform substantially better with dysarthric speech relative to speech in noise. The complex interplay between hearing loss and type of degradation will be discussed.

10:05–10:20 Break

10:20

4aNS7. Investigating individual susceptibility to the detrimental effects of background noise and reverberation in simulated complex acoustic environments. Kristi M. Ward (Northwestern Univ., 2240 N. Campus Dr., Rm. 2-381, Evanston, IL 60208, kmward@u.northwestern.edu), Z. Ellen Peng (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), Maryam Landi (National Ctr. for Physical Acoust. (NCPA), Univ. of MS, Oxford, MS), Andrew Burleson, Pamela Souza, and Tina M. Grieco-Calub (Northwestern Univ., Evanston, IL)

On a daily basis, individuals perform a variety of tasks in complex acoustic environments that contain background noise and reverberation. Previous research has demonstrated that, when tested in a laboratory setting, background noise and reverberation impair speech recognition and disrupt cognitive processing. These detrimental effects are especially pronounced for young children, older adults, and individuals with hearing loss. However, the specific environmental and individual factors that account for performance declines in the presence of background noise and reverberation, and whether these relations are generalizable to real-world complex acoustic environments, remains poorly understood. In the present study, children and adults performed speech recognition and speech comprehension tasks amidst background noise and reverberation in a state-of-the-art virtual sound room (ViSoR). ViSoR contains a Variable Room Acoustics System, which simulates the acoustic properties of real-world environments in a free-field test setting. Participants also completed standardized measures of attention, auditory working memory, and receptive vocabulary, which will be used to quantify the extent to which individual factors contribute to the observed changes in speech recognition and comprehension. Together, the findings from this study will provide additional insight as to the factors that underlie individual susceptibility to the detrimental effects of background noise and reverberation.

10:40

4aNS8. Effects of measured acoustic and other indoor environment factors on student achievement in K-12 classrooms. Kieren H. Smith (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Brigham Young Univ., Provo, Utah 84602, kierenhs@gmail.com), Ann Arthur, James Bovaird (Univ. of Nebraska-Lincoln, Lincoln, NE), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

Students can be affected by what they hear, see, and feel within the indoor classroom environment, whether or not it is cognizantly perceived by them. To understand how the indoor environment affects students' performance, measurements were logged in 220 K-12 classrooms in two Midwestern states during three seasons with two occupied days per season. Measurements of acoustics, indoor air quality, thermal comfort, and lighting were taken and a variety of metrics were calculated. For example, acoustic measurements provided reverberation time, clarity index, and equivalent sound levels during both active and inactive portions of the occupied school day. Achievement data in the form of percentile ranks on math and reading tests were also collected for the students who received instruction in each classroom. Structural equation models and multivariate linear regression models were utilized to analyze the effect of the indoor environment factors on student math and reading achievement. Significant findings, particularly in the area of acoustics, are presented and related to previous work in this area. [Work supported by the United States Environmental Protection Agency Grant Number R835633.]

11:00

4aNS9. Acoustic conditions for students' engagement in active learning classrooms. Shiva Hadavi and Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Saint-Catherine St. W, Montreal, Montreal, QC H3G 2W1, Canada, shiva.hadavi1988@gmail.com)

Educators have developed innovative teaching strategies in order to maximize learning outcomes in classrooms. Active learning classrooms are new learning spaces that facilitate the teaching strategies with enhanced students' engagement and collaborative discussions. Previous studies showed that the design of the space impacts on students' achievement. However, acoustic requirements of the active learning classrooms have not been investigated yet. This paper presents, thus, the acoustic conditions of the active learning classrooms located in Montreal. The acoustical parameters such as background noise, reverberation time, and speech transmission index in unoccupied conditions and occupied noise levels during learning activities are examined. The results are compared with conventional classrooms. Since active learning classrooms are more prone to acoustic deficiencies, the results of this paper can provide a better understanding of the acoustic design requirements for these spaces.

11:20

4aNS10. VR-based realistic assessment of annoyance for the noise of children running in apartment buildings. Hyun In Jo, Shahzad Ahmed, Hyun Wook Kim, Chun Ki Seo, and Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Hanyang University, Seoul, Seongdong-gu 133-791, South Korea, best2012@naver.com)

This study assessed annoyance response to temporal variations of sound energy of heavy floor impact in apartment buildings by using a virtual reality environment. As for a real moving sound source, the noise of a child running diagonally on the floor was reproduced by applying HRTF to the sound source of a single impact ball. Annoyance was assessed in the acoustic environment where visual information was provided by using HMD and directional information was given by using HRTF. A comparison scale using the equal-appearing interval (EQI) was adopted to reflect the sense of daily life. As a result, when visual information was provided in the VR environment, the annoyance of subjects decreased. On the other hand, directional information was given, the subjects became more sensitive to annoyance. Consequently, annoyance was more affected by the direction of a heavy moving sound source than the visual information. The VR-based environment of this study turned out to be realistic in a pyschoacoustic experiment.

11:40

4aNS11. Lombard effect in restaurant setting: How much would you spend to eat at this restaurant? Pasquale Bottalico (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S. 6th St., Champaign, IL 61820, pb81@illinois.edu)

The objective of this study is to determine the minimum level of noise in a restaurant that starts the Lombard effect, and how it relates to the perceived communication disturbance and the willingness to spend time and money for a meal. Twenty-eight participants were instructed to read a passage in the presence of restaurant noise from 35 dB(A) to 85 dB(A). As the noise level increased, participants began to be disturbed by the noise at 52 dB(A) and began to raise their voice at 57 dB(A). The willingness to spend time and money decreased starting at 52 dB(A).

THURSDAY MORNING, 8 NOVEMBER 2018 COLWOOD 1/2 (VCC), 8:30 A.M. TO 11:35 A.M.

Session 4aPA

Physical Acoustics and Biomedical Acoustics: Interactions of Sound Beams with Objects I

Likun Zhang, Cochair
University of Mississippi, 145 Hill Drive, Oxford, MS 38677

Grant Eastland, Cochair
System Acceptance & Operational Readiness, Naval Undersea Warfare Center Division Keyport, 610 Dowell St., Keyport, WA 98345

Invited Papers

8:30

4aPA1. Particle assembly and object propulsion using acoustic holograms. Kai Melde and Peer Fischer (Micro, Nano and Molecular Systems, Max Planck Inst. IS, Heisenbergstr. 3, Stuttgart, BW 70569, Germany, melde@is.mpg.de)

The contact-free manipulation of particles using ultrasound fields is an active field of research promising a number of applications. Conventional acoustic tweezers use strongly focused beams or higher order Bessel beams to provide a trap for single particles to be manipulated with. Other more mature methods use resonators to create elongated potential wells for collective particle trapping or separation. The resulting assemblies have a limited complexity, because the fields are highly symmetric. We recently introduced the acoustic hologram as an alternative way to create arbitrary ultrasound fields. In this talk, I will present two concepts, one for particle trapping and one for propulsion of objects, that have been enabled by this new method. The first is parallel assembly of microparticles at a surface in the shape of a projected acoustic image. Using a global trigger, these particles can be fused together to form a mechanically stable object. The second demonstration is a seemingly dynamic effect resulting from our static hologram. By defining the phase gradient (essentially the wave vector) along the water-air interface, it is possible to continuously propel objects along predefined trajectories. The necessary complexity to create such ultrasound fields with defined amplitude and phase distribution is easily managed using acoustic holograms.

8:50

4aPA2. Acoustic tweezer and its biomedical applications. K. K. Shung (Biomed Engr, Univ of S. Calif., 136 DRB, 1042 Downey Way, Los Angeles, CA 90089, kkshung@usc.edu)

Acoustic tweezer, the counterpart of optical tweezer in acoustics, was developed more than 10 years later than optical tweezer. It has been found to be capable of performing many tasks similar to optical tweezer but substantial differences also exist. Because of its larger wavelength, acoustic tweezer manipulates larger particles and yields larger forces in the nanonewton range. A number of biomedical applications have been studied including measuring intercellular forces and cellular mechanical properties. These results along with its physical principles will reviewed in this paper.

9:10

4aPA3. The development of dynamic holographic acoustic tweezers. Asier Marzo and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, University Walk, Bristol BS8 1TR, United Kingdom, b.drinkwater@bristol.ac.uk)

Radiation forces due to sound waves can be used to create acoustic tweezers that can trap and manipulate matter without contact. They provide unique advantages when compared to the more established optical tweezers, such as higher trapping forces per unit input power and the ability to trap a wide range of sample materials. This paper describes the development of dynamically reconfigurable holographic acoustic tweezers that can independently manipulate multiple millimetre-scale particles. We present hardware and algorithms that create converging acoustic radiation forces at multiple locations in space with an array of phase controlled emitters. We experimentally demonstrate a 40 kHz airborne ultrasonic system and manipulate up to 25 particles simultaneously. As the acoustic field is dynamically up-dated in real-time and manipulation speeds of up 40 mm/s are shown. When considered on the scale of a wavelength, this system has similar manipulation capabilities to optical tweezers. We show experimental results that demonstrate potential applications, e.g., for assembly processes both in the micro-metre and millimetric scale, as well as positioning and orientation of multiple object which could led to biomedical applications.

9:30

4aPA4. Applications of acoustic radiation force for material characterization and imaging. Mostafa Fatemi (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55905, fatemi.mostafa@mayo.edu) and Azra Alizad (Radiology, Mayo Clinic College of Medicine & Sci., Rochester, MN)

Acoustic radiation force has been used in many interesting medical and industrial applications in the past two decades. The fact that radiation stress can be confined to a small focal area and exerted remotely inside or on an object allows one to use this phenomenon in a variety of ways to explore properties of the object. Vibro-acoustography is a material characterization and imaging method that uses harmonic or impulsive acoustic radiation force to stimulate the targeted material and receives the resulting acoustic response. The output acoustic signal is then used to extract important information about the object. Biomedical applications of vibro-acoustography and its closely related methods include imaging, flow detection, and tissue characterization. Other applications include non-destructive evaluation/testing of materials, modal analysis of small structures, and microscopy. This talk presents an overview of vibro-acoustography and its closely related techniques over the past 20 years. Various biomedical and industrial applications will be presented. The talk will conclude with a discussion on potential future applications of this technology.

9:50

4aPA5. Compact selective tweezers based on focalized acoustical vortices and spiraling interdigitated transducers. Michael Baudoin, Jean-Claude Gerbedoen, Antoine Riaud, Olivier Bou Matar (IEMN, Univ. Lille, CNRS, Centrale Lille, ISEN, Univ. Valenciennes, UMR 8520, IEMN, Ave. Poincaré, Villeneuve d'Ascq 59652, France, michael.baudoin@univ-lille1.fr), Nikolay Smagin (IEMN, Univ. Lille, CNRS, Centrale Lille, ISEN, Univ. Valenciennes, UMR 8520, Valenciennes, France), and Jean-Louis Thomas (INSP, Sorbonne Universités, CNRS UMR 7588, Paris, France)

With the emergence of regenerative medicine, cell printers, and labs on chips, the contactless selective manipulation of microscopic objects such as particles, cells, or drops has become a key feature. To complete this task, acoustic tweezers appear as a tremendous alternative to their magnetic and optical counterpart. Indeed, they do not require pre-tagging of the manipulated object and they enable particles trapping with forces several orders of magnitude larger than optical tweezers at same input power. Recently, Baresh *et al.* [*Phys. Rev. Lett.*, **116**, 024301 (2016)] demonstrated the selective 3D manipulation of particles with a specific class of waves called acoustical vortices. Nevertheless, such manipulation was achieved with a complex transducer array coupled with a high end programmable electronics. This system is cumbersome, not compatible with microscopes and hardly miniaturizable. To overcome these difficulties, our team developed new tweezers [Riaud *et al*., *Phys. Rev. Appl.* **7**, 024007 (2017)] based on spiraling interdigitated transducers (IDTs), some electrodes sputtered at the surface of piezoelectric substrates patterned by photolithography. The shape of the electrodes encodes the phase of the field like a hologram. For applications, these tweezers have many attractive features: they are selective, flat, easily integrable, and compatible with disposable substrates.

10:10–10:25 Break

10:25

4aPA6. Phase-shift based expansions for material and frequency dependence of scattering and of radiation forces. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

When considering the scattering of sound and radiation forces for spheres, it has historically been helpful to gain a good understanding of situations lacking dissipation. In that case the linear acoustic scattering by spheres is characterized by real partial-wave phase shifts. At low frequencies it is possible to obtain simple expansions showing the dependence of each phase shift on material properties and on frequency [P. L. Marston and L. Zhang, *J. Acoust. Soc. Am.* **141**, 3042–3049 (2017); P. L. Marston, J. Acoust. Soc. Am. 142, 1167–1170 (2017); P. L. Marston, J. Acoust. Soc. Am. 142, 3358–3361 (2017)]. Those expansions can then be used to describe the frequency and material dependence of scattering and radiation forces for standing and traveling waves at low frequencies beyond the usual Rayleigh-scattering approximation. While spherical Bessel and Hankel functions are used in the derivation, the final expansion coefficients are expressed using algebraic functions. The derivation of the simplified series expansion for the radiation force in the traveling wave case is subtler than the standing wave case. Well-known limiting expressions are recovered in both cases. Some applications to spheres in acoustic beams are also known. [Work supported by ONR.]

10:45

4aPA7. Dipolar and quadrupolar mode dissipation of spherical probes spinning in vortex beam acoustical tweezers. Diego Baresch (Dept. of Chemical Eng., Imperial College London, London SW7 2AZ, United Kingdom, diego.baresch@upmc.fr), Régis Marchiano (Institut Jean Le Rond D'Alembert, UPMC, Paris, France), and Jean-Louis Thomas (Institut des NanoSci. de Paris, UPMC, Paris, France)

In this work, the possibility of simultaneously trapping and rotating single polystyrene beads, or clusters, against gravity with an ultrasonic vortex beam is demonstrated. The induced rotation of a single particle is compared to a torque balance model accounting for the acoustic response of the particle. Two dominating dissipation mechanisms of the acoustic orbital angular momentum responsible for the observed rotation are examined. The first takes place in the bulk of the absorbing particle, while the second arises as the angular momentum flux is dissipated in the viscous boundary layer surrounding the particle. Importantly, even in the long-wavelength (Rayleigh) regime, it is crucial to fully model the dissipation of the dipolar and quadrupolar vibrational modes of the particle in a slightly viscous fluid such as water. Further results suggest that, while the induced outer (Eckart) rotational flow can be neglected in water, it can play an important role in other viscous and complex fluids.

Contributed Papers

11:05

4aPA8. Measurement and calculation of lateral trapping strength of focused beams generated by a two-dimensional ultrasound array. Mohamed A. Ghanem (Aeronautics and Astronautics Eng., Univ. of Washington, Box 352400, Seattle, WA 98195-2400, mghanem@uw.edu), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Seattle, Washington), 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), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

This study is aimed at developing a method to controllably manipulate kidney stones in the human body using a single ultrasound source. Calculation and measurement of the lateral force on a sphere using various acoustic beams (vortex and radially-varying phased beams) were compared. A 256-element, 1.5-MHz array was used to synthesize the beams. Spheres of 1–6 mm ($6 < ka < 20$) diameter made of glass, ceramic, or brass were positioned on an acoustically matched platform at the focus rigidly attached to the transducer by a frame. The transducer and platform were rotated to the angle at which the trapped sphere fell. The acoustic power was <9 W and was adjusted by the duty cycle (10–60%) to control the range of the trapping angle. Maximum temporal average intensity I_{SPTA} was 46.5 W/cm^2. The acoustic force was calculated numerically as in Sapozhnikov and Bailey [JASA, 133, 616 (2013)] and the angle calculated with static force equilibrium equation. Good agreement between calculation and measurement was observed, with an average error in angle measured of 11.3%. The maximum lateral forces were 80% of the axial radiation force and 52% the gravitational force. [Work supported by NIH P01-DK043881, K01-DK104854, R01EB7643, and RBBR 17-02-00261.]

11:20

4aPA9. Acoustic non-diffracting tractor beams with lateral trapping. Xudong Fan (Univ. of MS, University, MS) and Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

In spite of the recent exploration of non-diffracting tractor beams that pull a particle by nonconservative forces, the pulling over a long distance is not realistic without the simultaneous trapping in the lateral direction by gradient forces. Trapping of a small particle by both ordinary and vortex Bessel beams is examined in the parameter space of mass density and compressibility. The results reveal beam parameters for stable trapping. It is found that under certain conditions the trapping is independent on paraxiality parameter of the beam. The findings are validated by calculations for a variety of objects. Material parameters for the simultaneous trapping and pulling of a small particle by an ordinary Bessel beam is identified. These findings pave the way for experimental realization of stable acoustic tractor beams for a small particle.

Session 4aPP

Psychological and Physiological Acoustics: Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations I

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Chair's Introduction—9:15

Invited Papers

9:20

4aPP1. Challenges and opportunities in bridging behavior, physiology, and anatomy in translational hearing research. Amanda Lauer, Ye-hyun Kim, and Katrina Schrode (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

The Acoustical Society of America has a long, rich history of physiology research aimed at understanding the neural mechanisms underlying hearing and hearing dysfunction. In particular, this research has included pivotal studies in animal models that reveal the neurobiology of hearing and hearing disorders. More recently, research in animal models and research in human subjects have been considered in completely separate conferences or sessions. This situation presents a barrier to communicating important scientific information and fostering collaborations that bridge the gap between research in animal models and human listeners. It is crucial that the two types of research interface and inform one another to maximize translational potential and converge on fundamental knowledge. Experimental manipulations can be performed in non-human animals that cannot be performed in humans, but human and non-human animals can also be tested using similar techniques to increase translatability. Examples of mouse models tested using detailed analysis of synapse-level changes, clinically relevant evoked potentials, and behavioral tests will be presented. Areas for mutually beneficial interdisciplinary investigation will be discussed with the intention of inspiring new lines of communication between researchers focused on non-human animal and human subjects research.

9:40

4aPP2. Importance of translational research in audiology. Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu)

Students, postdocs, and junior research-faculty in audiology all face an interesting challenge: which conferences should I attend, and can I afford the time and likely financial burden associated with attending several meetings in one year? As research shifts towards translational topics, it becomes difficult to remain siloed within our respective fields. It is increasingly important for clinical researchers in audiology to communicate with specialized medical doctors, physiologists, engineers, psychoacousticians, etc. An example of such technological advancements that may contribute to the field of audiology are neuroimaging techniques and signal processing capabilities. For instance, enhanced imaging techniques allow for mapping the auditory pathway for different spatial cue information and auditory prostheses are now capable of retaining or supplementing binaural information through signal processing. But how many patients visit an audiologist with complaints of sound localization ability? In this example, it is critical for clinical researchers to be aware of physiological and technological advancements in order to appropriately translate spatial hearing research to the clinic. Here, I will discuss the benefit of clinicians attending translational auditory conferences by highlighting binaural hearing research and why attendance at translational conferences such as the Acoustical Society of America is essential for clinical research in audiology.

10:00

4aPP3. From behavior to physiology and back again: The role of auditory cortex in vocal production and control. Steven Eliades and Joji Tsunada (Otorhinolaryngology: Head and Neck Surgery, Univ. of Pennsylvania, 3400 Spruce St., 5 Ravdin, Philadelphia, PA 19104, seliades@pennmedicine.upenn.edu)

Vocal communication plays an important role in the lives of both humans and many animal species. Ensuring accurate communication, however, requires auditory self-monitoring to control vocal production and rapidly compensate for errors in vocal output. Despite the importance of this process, the underlying neural mechanisms are relatively unknown. Previous work has demonstrated that neurons

in the auditory cortex are suppressed during vocal production, while simultaneously maintaining their sensitivity to vocal feedback, suggesting a role in auditory self-monitoring. The behavioral role of auditory cortex in vocal control, however, remains unclear. We investigated the function of auditory cortical activity during vocal self-monitoring and feedback-dependant vocal control in marmoset monkeys. Using real-time frequency-shifted feedback during vocalization, we demonstrate that marmosets exhibit rapid compensatory changes in vocal production, a feedback-dependent behavior that is predicted by the activities of neurons in auditory cortex. We further establish the role of auditory cortex in vocal control using electrical microstimulation to evoke rapid changes in produced vocalizations. These findings suggest a causal role for the auditory cortex in vocal self-monitoring and feedback-dependent vocal control, linking mechanisms of production and perception, and have important implications for understanding human speech motor control.

10:20

4aPP4. Blast exposure impairs sensory gating: Evidence from measures of acoustic startle and auditory event-related potentials. Melissa A. Papesh and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 4000 SW 91st Court, Portland, OR 97225, Melissa.Papesh@va.gov)

Many military Veterans who have been exposed to high-intensity blast waves suffer from traumatic brain injury (TBI), resulting in chronic auditory deficits despite normal hearing sensitivity. The current study examined the neurological cause of this chronic dysfunction by testing the hypothesis that blast exposure leads to impaired filtering of sensory information at brainstem and early cortical levels. Groups of blast-exposed and non-blast-exposed participants completed self-report measures of auditory status, auditory perceptual tasks, and physiological measures of sensory gating, including prepulse-inhibition and habituation of the acoustic startle reflex and electrophysiological assessment of a paired-click sensory gating paradigm. Blast-exposed participants showed significantly reduced habituation to acoustic startle stimuli and impaired filtering of redundant sensory information at the level the auditory cortex. Linear regression analyses revealed that poorer sensory gating at the cortical level was primarily influenced by a diagnosis of TBI, while reduced habituation was primarily influenced by a diagnosis of posttraumatic stress disorder. A statistical model was created including cortical sensory gating and habituation to acoustic startle, which strongly predicted performance on a degraded speech task. These results support the hypothesis that blast exposure impairs central auditory processing via impairment of neural mechanisms underlying habituation and sensory gating.

10:40–10:55 Break

10:55

4aPP5. Subcortical neural correlates of behavior in the auditory system. Ramnarayan Ramachandran (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 111 21st Ave. South, Wilson Hall 065, Nashville, TN 37240, ramnarayan.ramachandran@vanderbilt.edu)

One of the biggest open questions in neuroscience is the relationship between neuronal activity and behavior. Little is known in the auditory system about the nature of the relationship between stimulus encoding and perception. To shed light on this topic, we recorded responses of single neurons in the central nucleus of the inferior colliculus (CNIC) and cochlear nucleus (CN) of macaques that were trained to detect tones or vowels during behavioral performance. Using signal detection theoretic methods, we determined that neurometric tone and vowel detection thresholds based on rate measures in CNIC matched behavioral thresholds, while thresholds of CN neurons were higher than behavior. Additionally, the response magnitudes of a majority of the CNIC neurons varied significantly depending on the behavioral accuracy, compared to a much smaller proportion of CN neurons. Analysis of temporal patterns of response (e.g., phase locking) suggests that neurometric thresholds of all CN and most CNIC neurons matched behavioral thresholds during vowel detection. Additionally, every CNIC and CN neuron showed significant response pattern differences based on behavioral accuracy. These results suggest that behaviorally relevant coding early in the pathway is temporal in nature and is transformed to more rate based measures at higher levels of processing.

11:15

4aPP6. Suprathreshold hearing in middle age and relationship to cochlear synaptopathy. Hari M. Bharadwaj, Brooke Flesher, Alexandra Mai, Kelsey Dougherty, Jennifer M. Simpson, and Michael G. Heinz (Purdue Univ., 715 Clinic Dr., Lyles-Porter Hall, West Lafayette, IN 47907, hbharadwaj@purdue.edu)

Animal models have demonstrated that the afferent synapses and nerve terminals innervating the cochlea are vulnerable to damage from acoustic overexposure and aging. This synaptopathy can occur without hair-cell loss, or more severely when accompanied by permanent audiometric shifts. In humans, postmortem temporal bone studies have shown that cochlear synaptopathy occurs throughout adulthood, decades before audiometric loss nominally occurs. However, effective non-invasive assays of synaptopathy have yet to be established in humans (or genetically heterogeneous animal cohorts). Moreover, whether synaptopathy contributes to age-related temporal perception deficits is debated. We are currently studying young and middle-aged humans with clinically "near-normal" audiograms using physiological and perceptual measures. Preliminary results suggest that although high-frequency audiometric shifts occur with age as previously known, middle age per se is associated with physiological effects consistent with cochlear synaptopathy. Further, we find that the effects of age and threshold elevation can oppose and obscure one another on certain perceptual measures. Finally, with speech-in-noise measures designed to degrade high-frequency envelope cues, perceptual deficits associated with middle age per se are apparent and consistent with synaptopathy. These results will be discussed in light of the ongoing debate about the prevalence and consequences of synaptopathy in humans.

11:35

4aPP7. The role of central processing in modulation masking release. Nima Alamatsaz and Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd, Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu)

When background sound is present, hearing impaired (HI) individuals and cochlear-implant (CI) listeners typically are worse at hearing out target sound as compared to normal-hearing (NH) listeners. This perceptual deficit occurs both when the background consists of noise that fluctuates over time (“modulated”) and for stationary background noise (“unmodulated”). In addition, the difference in thresholds between tone detection in modulated and unmodulated noise, referred to as modulation masking release (MMR), is much reduced or absent in HI and CI as compared to NH. Both peripheral and central processing mechanisms contribute to MMR. We previously showed that central MMR is reduced in human CI listeners, and that sound deprivation reduces central MMR in Mongolian gerbils. Here, we began to explore the neurobiological basis of central MMR. NH gerbils were trained to hear out target tones (1 kHz) in modulated (10-Hz rectangularly gated) versus unmodulated bandlimited background noise, and chronically implanted with recording electrodes in core auditory cortex. Neural discharge was analyzed as a function of the broadband energy ratio between target and background sound to determine how different types of background sound affect neural information transmission in awake behaving gerbil. Preliminary results will be discussed in the context of how hearing loss may affect central MMR.

THURSDAY MORNING, 8 NOVEMBER 2018 — UPPER PAVILION (VCC), 9:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Speech Production (Poster Session)

Laura L. Koenig, Chair
Haskins Labs and Long Island University, 300 George Street, New Haven, CT 06511

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

4aSC1. Prominence-marking mechanisms in emotional speech. Marcus Vinicius Moreira Martins (Faculdade de Letras, Universidade Federal de Minas Gerais, Avenida Antônio Carlos, 6627, Belo Horizonte, Minas Gerais 31270901, Brazil, marcusvmmartins@gmail.com) and Waldemar Ferreira-Netto (Departamento de Letras Clássicas e Vernáculas, Universidade de São Paulo, São Paulo, São Paulo, Brazil)

The aim of this work is to investigate the prominence-marking mechanisms in emotional speech, particularly in Brazilian Portuguese intonational system. Three professional actresses interpreted a text in three conditions: anger, sadness and neutral, in a total of 130 sentences for each condition (n = 390). Others tests established the minimum value for a prominence mark is +3 semitones from the mean fundamental frequency (F0). In the first analysis, it is observed the distribution of prominences frequencies (F0) is normal for neutral, but non-normal in the other two cases (skewness; anger = -1.45 and sadness = -0.5) and the difference between distributions is significant in all cases. In the rage and neutral conditions, the prominence is marked consistently 5 semitones above fundamental frequency mean, non-random, however in sadness the prominence-marking mechanism is almost random. A second frequency distribution test revealed the difference between distributions is significant between anger and other conditions ($p < 0.05$), but it is not significant between sadness and neutral speech ($p > 0.05$). These data suggest that the distribution of frequencies in the pitch range combined with an analysis of the prominences may be a relevant criterion to distinguish the sadness from other emotions, especially in automatic recognition process.

4aSC2. Neighborhood-conditioned coarticulation effects in French listener-directed speech. Rebecca Scarborough (Linguist, Univ. of Colorado, 295 UCB, Boulder, CO 80309, rebecca.scarborough@colorado.edu), Cécile Fougeron (LPP, CNRS/Université Sorbonne Nouvelle, Paris, France), and Luciana Marques (Linguist, Univ. of Colorado, Boulder, CO)

Words from dense phonological neighborhoods (Hi-ND words) are realized with greater vowel hyperarticulation and increased coarticulation in English, relative to words from sparser neighborhoods (Lo-ND words) [e.g., Wright 2004]. Here, the relation between coarticulation and neighborhood density is investigated for French by looking at patterns for anticipatory and carryover nasal coarticulation, coronal and uvular CV coarticulation, and V-to-C rounding coarticulation. Twelve native French speakers (northern France) produced 82 disyllabic words of French containing a context for one of these 5 types of coarticulation in phonetically similar high-low ND pairs. Test words were produced to a real listener who had to transcribe them in the instructed positions on a grid. F1, F2, and A1-P0 (a spectral measure of nasality) were measured at 5 timepoints across each test vowel. In addition to having more hyperarticulated (peripheral) vowel midpoints, Hi-ND words showed greater anticipatory nasality and stronger coronal transitions in appropriate contexts than Lo-ND words. Carryover nasality and uvular and rounding coarticulation did not differ across NDs. These results can be interpreted in light of the relation between coarticulation and contrast and the role these effects may serve in making Hi-ND words, which are subject to greater lexical competition, easier to perceive.

4aSC3. Speech-like movements emerge from simulated peri-oral muscle activation space without neural control. Jonathan M. de Vries (Interdisciplinary Studies Graduate Program, Univ. of Br. Columbia, 270, 2357 Main Mall, Vancouver, BC V6T 1Z4, Canada, devriesj@alumni.ubc.ca), Ian Stavness (Computational Sci., Univ. of SK, Saskatoon, SK, Canada), Sid Fels (Elec. And Comput. Eng., Univ. of Br. Columbia, Vancouver, BC, Canada), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Muscle synergies have been proposed as a solution to the degrees of freedom problem inherent in the biomechanics of speech production [Gick & Stavness, 2013, *Front. Psych, 4*, 977]. However, the majority of experiments involving the extraction of muscle synergies prove theoretically insufficient to determine whether such synergies are of neural origin or simply reflect the lower dimensionality of an under-sampled biomechanical/ neural task space [Kutch & Valero-Cuevas, 2012, *PLoS computational biology*, *8*(5), e1002434]. Therefore, to what extent biomechanics of the human vocal tract may constrain the "ecological state space" of a speaker during locution remains uninvestigated. As a proof of concept, we created a simplified version of the peri-oral region using FEM modeling in a physics-based simulator (ArtiSynth [Lloyd *et al*., 2012, *Soft tissue biomech…surgery,* pp. 355]). Systematic simulations enabled us to model the full kinematic/biomechanical space. Visualization of the resulting biomechanical state space using t-SNE [van der Maaten & Hinton, 2008, *J.MLR*., *9,* 2579] illustrates that speech-like movements (e.g., lip rounding or spreading) emerge as self-organizing structures (muscle synergies) critically without a direct neural controller. These results are discussed in the context of current and future explorations of motor control of speech. [Funded by NSERC.]

4aSC4. Frequency effects in palatalization: Articulatory evidence from English. Jae-Hyun Sung (Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, jsung@yonsei.ac.kr)

Lexical frequency and its influence on speech production have been widely acknowledged across languages and varieties, and frequency effects are so prevalent in various phonological and coarticulatory phenomena. Previous experimental studies have reported that high lexical frequency leads a greater chance of overt coarticulation (for example, more frequent overt palatalization of /d/ in *would you* (high-frequency) than *bad you* (low-frequency)). This study tests for articulatory evidence of frequency effects in palatalization by examining palatalized consonants resulted by lexical and post-lexical palatalization in English using ultrasound imaging. Comparisons of tongue contours of palatalized consonants produced by 12 native speakers of American English show that differences in lexical frequency are not directly linked to articulatory gestures. That is, speakers do not necessarily produce a greater degree of palatalization (i.e., tongue contour closer to the palate) in high-frequency words or phrases. Moreover, while most speakers maintain some degree of articulatory contrast in high- vs. low-frequency contexts, the way speakers differentiate palatalized consonants from high- vs. low-frequency items is highly individualized. The findings from this study are in line with individual variation in coarticulation, and merit further exploration in the role of lexical frequency in speech production.

4aSC5. Lexically conditioned phonetic variation: A test with the voicing contrast in Japanese. Keiichi Tajima (Dept. of Psych., Hosei Univ., 2-17-1 Fujimi, Chiyoda-ku, Tokyo 102-8160, Japan, tajima@hosei.ac.jp), Kiyoko Yoneyama (Dept. of English, Daito Bunka Univ., Tokyo, Tokyo, Japan), and Mafuyu Kitahara (Dept. of English Studies, Sophia Univ., Tokyo, Japan)

Many lexical factors have been shown to influence phonetic realization of words. For example, studies have shown that voice onset time (VOT) of word-initial stops is shorter in high-frequency words than in low-frequency words, and is longer in words that form a voicing minimal pair, e.g., *cod-god*, than in words that do not, e.g., *cop-*gop*. The present study begins to ask whether such lexically conditioned phonetic variations are language-general, by examining productions of words in Japanese. The stimuli were two-mora Japanese minimal pairs contrasting in word-initial /k/ vs. /g/, half of which were real words, e.g., /kara/, while the other half were similar-sounding nonwords, e.g., */kapa/. Furthermore, half of the items had a lexical competitor contrasting in voicing, e.g., /kara/-/gara/, while the other half did not, e.g., /kana/-*/gana/. The stimuli were split so that each participant read only one member of each minimal pair. Twenty-four native Japanese speakers read the target items interspersed with filler items. Results showed opposite trends from those previous reported. Specifically, VOT for /k/-initial words was *longer* for (high-frequency) real words than for (low-frequency) nonwords, and was *shorter* for words that had a lexical competitor than for words that did not. [Work supported by JSPS.]

4aSC6. Vowel formant trajectories in naturalistically produced loud speech. Susanne Fuchs (Leibniz-Zentrum Allgemeine Sprachwissenschaft, Berlin, Germany) and Laura L. Koenig (Haskins Labs and Adelphi Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu)

Louder speech, relative to typical speech, may show changes in addition to the expected increased SPL, such as higher values of the first formant (F1), longer durations, and higher articulatory velocities. Most past work has assessed vowel midpoints only. To what extent can any formant differences be observed across the full time course of the vowel? Here we evaluate formant trajectories in loud and conversational speech. Adult female speakers of German produced acronyms containing /a: i: u:/, and words that put various vowels in a bilabial$_{\text{alveolar}}$ context. Greater loudness was elicited by increasing speaker-experimenter distance. Preliminary analyses extracting values at 25%, 50%, and 75% of the vowel duration show that formant trajectories in normal and loud speech are usually quite parallel throughout the vowel, particularly for high and tense vowels (where loudness differences are minimal overall). In some cases normal-loud differences may increase from the 50% to 75% time points. In general, it does not appear that articulatory/acoustic differences between loud and normal speech are restricted to vowel midpoints. We will also carry out analyses over shorter time intervals to evaluate very early and late regions in the vowels.

4aSC7. Differences in clear speech strategies across read and spontaneous speech produced in interactive tasks for young and older adults. 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 clear speech strategies vary according to the demands of the task used for eliciting clear speech. Spontaneous speech involves greater planning than sentence reading and this might particularly affect talkers who find communication more effortful. 77 talkers (24 young and 53 older adults) carried out two collaborative tasks with a conversational partner in two listening conditions. The "read" task involved reading a sentence to the partner who repeated it back; the "spontaneous" task involved completing the diapix problem-solving task. Partners either heard each other easily (NORM) or communicated with the partner having a simulated profound loss, naturally eliciting clear speech (CLEAR). The percentage relative change between NORM and CLEAR was calculated for articulation rate, long-term average spectrum, fundamental frequency, vowel formant ranges. Analyses focus on task and age effects. There were significant task [$F(1,75) = 10.27; p = 0.002, \eta 2 = 0.886$] and task by measure effects [$F(5,375) = 17.97; p<0.001, \eta 2 = 0.193$], with a greater reduction in articulation rate for read (M = -27.3%) than spontaneous (M=-9.2%) speech, $p<0.001$. Task effects did not vary significantly with talker age. In summary, the use of read speech may over-emphasise the role of articulation rate as a clear speaking style strategy but this was the case for young and older talkers.

4aSC8. Do speakers maintain clear speech throughout natural conversation? Dae-yong Lee and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 161 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, daeyongl@uoregon.edu)

Previous studies have shown that speakers tend to speak more clearly when they speak with a listener in adverse listening conditions in naturalistic settings. Most previous studies assumed that when speakers speak more clearly, they maintain the style throughout the conversation. However, a clear speech style deviates from speakers' conversational speaking style, and extra effort is required in order to produce clear speech. Therefore, it is possible that speakers begin a conversation in a clear speaking style and

revert back to a conversational style at some point during the conversation. This study aims to examine whether native English speakers use and maintain a clear speaking style while participating in a conversation with non-native speakers in naturalistic settings. Native English listeners will respond to sentences from naturalistic conversations. The sentences are from either a conversation between two native English speakers or a conversation between a native English speaker and a non-native English speaker. Intelligibility (typing what the listeners heard) and comprehensibility (rating how easy it is to understand the sentences) are measured to examine the usage of clear speech throughout the conversations. The results of this study will provide a better understanding of the circumstances under which speakers use clear speech.

4aSC9. Geminated liquids in Japanese. Maho Morimoto (Linguist, Univ. of California, Santa Cruz, 1156 High St., Santa Cruz, CA 95064, mamorimo@ucsc.edu) and Tatsuya Kitamura (Faculty of Intelligence and Informatics, Konan Univ., Kobe, Hyogo, Japan)

Liquid geminates are uncommon and disfavored in Japanese native phonology. However, instances of emphatic expressions and loanwords from languages with liquid geminates such as Italian or Arabic suggest that they are not impossible. The current study examines the durational, acoustic, and articulatory properties of geminated liquids in Japanese to obtain further insights into the nature of /r/ in Japanese and the process of gemination in general. We report on a production experiment whereby eight native speakers of Japanese pronounced mimetics of the form CVCVCVCV with and without emphatic gemination (e.g., kirakira > kir:akira “shiny”). In addition to the audio, tongue movements were recorded using the EMA technology. Preliminary durational analysis suggests that singleton-geminate ratio is about 1:3, which is slightly larger than the ratios previously reported for geminated obstruents. We explore what articulatory strategies speakers employ to lengthen the liquid consonant whose prototypical singleton production is [ɾ].

4aSC10. Final devoicing in Singapore English. Daryl Chow (English Lang. & Lit., National Inst. of Education, Singapore, Singapore, Singapore) and Viktor Kharlamov (Lang., Linguist & Comparative Lit., Florida Atlantic Univ., 777 Glades Rd, CU-97, Ste 280, Boca Raton, FL 33431, vkharlamov@fau.edu)

This study investigates the voicing contrast in word-final obstruents in Singapore English. Previous auditory accounts have suggested that word-final obstruents undergo complete devoicing in Singapore English and become identical to their voiceless counterparts, including neutralization of differences in adjacent vowel duration (among others, Bao 1998, Wee 2008). On the basis of acoustic data from the NIESCEA corpus (Low 2015), we examine the extent to which voicing-related differences are neutralized when reading wordlists, sentences and passages as well as in unscripted conversations. Results suggest that neutralization is phonetically incomplete in Singapore English, with underlying voicing being recoverable from such parameters as closure voicing ratio and durations of consonants and adjacent vowels. These findings are in line with the results for other devoicing languages, such as German and Russian, that also show incomplete neutralization of the voicing contrast in final obstruents (e.g., Roettger *et al*. 2014, Kharlamov 2014).

4aSC11. Positional allophony in ejective stops: A case study of Georgian. Chloé Gfeller (UFR Linguistique, Université Paris Diderot - Paris 7, 5 rue Thomas Mann, UFR Linguistique - à l’attention de Mme Chitoran, Paris cedex 13 75205, France, chloe.gfeller@gmail.com)

Many phonetic studies have dealt with ejective stops, but allophonic differences between initial and intervocalic positions have not been investigated. Ejectives have been typologically classified as either “stiff” or “slack,” depending on their glottal tension (Kingston 1985). Since stops in general tend to show positional acoustic differences superficially similar to those induced by differences in glottal tension (notably differences in VOT duration), we ask whether positional allophony of ejectives manipulates glottal tension. In an acoustic study of Georgian ejectives, we observed longer mean VOTs in initial position for labial (28 ms vs 23 ms) and coronal ejectives (33 ms vs 25 ms), consistent with increased glottal tension. Additionally, in a subset of cases, the vowel following an intervocalic ejective is completely glottalized, consistent with decreased glottal tension in intervocalic position. However, we found no significant difference overall in the mean duration of glottalized vowel phonation after ejectives between initial and intervocalic positions, notably because, in the other intervocalic tokens, the duration of vowel glottalization is comparable to that in initial position. This suggests that the observed positional differences in VOT in Georgian ejective stops are not reducible to differences in glottal tension.

4aSC12. The role of vertical larynx movement in the tense-lax contrast of Seoul Korean stops across phrasal position. Yoonjeong Lee and Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, GFS 301, Los Angeles, CA 90089, yoonjeol@usc.edu)

In contemporary Seoul Korean (SK), phrasal tone patterns are co-active with tone patterns that distinguish tense and lax stops. Utilizing real-time MRI, the current investigation of the SK stops elucidates the articulatory synergies that distinguish tense versus lax and how they function within SK’s Accentual Phrase (AP) prosodic system. The f0 and corresponding larynx height values were measured from sequences containing the three oral stops of SK placed in AP-initial and AP-internal prosodic positions. Larynx height was obtained by finding the intensity centroid within a locally selected region in the MR image around the larynx. We find a strong *positive* correlation between f0 and larynx height, indicating that vertical larynx movement is engaged in some tonal manipulation. We confirm the *categorical* consonant effect on f0 AP-initially (tense >> lax), compared to a *gradient* effect AP-internally (tense >= lax). In AP-initial position, the larynx height results mirror the f0 results, suggesting that the f0 goals associated with tenseness result in large part from a vertical larynx position manipulation. In AP-*internal* position, however, larynx height does not differentiate tense versus lax consonants, suggesting that some other articulatory activity, such as vocal fold stretching, is responsible for its f0 variations. [Work supported by NIH.]

4aSC13. The diphthongs of the acrolectal Kenyan English spoken by the Black Indigenous Kenyans. RUTH W. IRUMBI (Commun., Lang. and Linguist, Pan Africa Christian Univ., Nairobi, Nairobi 01000, Kenya, wrndungu@gmail.com), Mathew K. Karia (Special Needs, Kenyatta Univ., Nairobi, Nairobi, Kenya), and Joshua M. Itumo (English Lang. & Linguist, Kenyatta Univ., Nairobi, Nairobi, Kenya)

Kenya has three main varieties of English: the variety spoken by white Kenyans (WhKE), the acrolectal Kenyan English spoken by the Black Indigenous Kenyans (BIKE), and the mesolectal varieties, which are ethnically marked (Kioko & Muthwii, 2004; Hoffmann, 2010; Njoroge, 2011).There have been calls to describe the acrolectal variety of English .This paper documents the BIKE diphthongs based on data from a research carried out in 2016. The participants, seven males and seven females, were university lecturers who were purposively selected from the three major language groups in Kenya (Bantu, Cushitic and Nilotic). These respondents read “*The Boy Who Cried Wolf*”, a passage adapted for IPA English phonemic analyses. Four tokens for each of the eight RP diphthongs were sampled giving us 448 tokens. The recorded audio files were converted into *.wav* files for acoustic analysis using *Praat* (Version 6.0.05). The findings presented herein, show that KenE has six diphthongs: / ai/, /oi/, /ua/, /ia/, /ea/, /ua/ and /au/, which are acoustically different from the eight RP diphthongs. In this paper, we explain how the BIKE diphthongs differ from the RP ones; describe how they cluster based on the gliding; and also map them in the vowel trapezium.

4aSC14. Canadian French rhotics and their continued change. Justin T. Craft and Tamarae Hildebrandt (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, juscraft@umich.edu)

In previous decades, Canadian French has undergone a sound change throughout Western Quebec where the apical trill [r] has moved to a production using the tongue dorsum, often being produced by speakers as a velar fricative [x, ɣ], uvular trill [ʀ], or uvular fricative [χ, ʁ] (Sankoff and Blondeau 2007, 2010). This change is argued to have been primarily driven by younger female speakers adopting the dorsal production, but change is also reported across the lifespan of older speakers. Discrepancies about whether variation in rhotic productions is the result of allophony from syllable position (Sankoff and Blondeau 2010) or whether variation is socially constrained (Milne 2012) also exist in the literature. This study presents an analysis of rhotics in harmonic words as produced by 9 speakers from Montreal, Quebec. We report preliminary results highlighting the acoustic differences in production (measuring center of gravity, F1, & F0) between speakers of different genders. These results support the hypothesis that variation is socially constrained, in addition to lending support to early reported hypotheses about the role of women as a driving force of this sound change (Sankoff *et al.* 2001).

4aSC15. Regional variation in West Yorkshire filled pauses: Implications for forensic speaker comparisons. Erica Gold, Sula Ross, and Kate Earnshaw (Dept. of Linguist and Modern Lang., Univ. of Huddersfield, Queensgate, Huddersfield HD1 3DH, United Kingdom, e.gold@hud.ac.uk)

In the current linguistic literature, West Yorkshire (a county in Northern England) has received relatively little commentary, as it is often overshadowed by other bigger regions and cities like Manchester or Newcastle-upon-Tyne. However, in forensic phonetics, literature on regional variation is often vital to forensic casework. Sociophonetic studies aid forensic phoneticians in making judgments regarding whether speaker characteristics are typical of a region or not. For both forensic and sociophonetic motivations, this paper begins to look at the variation present in West Yorkshire by analyzing variation in filled pauses across speaking styles from three boroughs within West Yorkshire (Bradford, Kirklees, and Wakefield) from the West Yorkshire Regional English Database (WYRED; Gold *et al.*, 2016). This paper analyses 60 speakers from WYRED. All speakers are male, aged 18-30, and English is their first and only language. This study measures the vocalic portion of all "uh" /V/ and "um" /V + N/ productions in over 8,000 tokens across four different tasks. Filled pauses were manually segmented in Praat and F1, F2, and F3 midpoints were extracted. Results suggest that filled pauses may be influenced by speaking style, but more importantly the three regions exhibit some significant differences in their filled pause realizations.

4aSC16. Distributional patterning of reflex cough during telephone conversations. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu)

Reflex cough and speech are two behaviors that make use of the same end effectors—the respiratory and vocal tract subsystems. How do talkers engaged in conversation resolve conflicts between the communicative necessities involved in speech planning and their physiological reflexes? Coughing is a complex reflex arc triggered in the airway that leads to an urge to cough and, at some latency, the deployment of a coordinated vocal-respiratory response to that urge. However, research on factors that influence urge-cough latency is limited. Hypothesizing that talkers may actively delay coughing, this present study examines 200 instances of reflex cough from the Switchboard and CallHome corpora of spontaneous, dyadic telephone speech. Coughing events in both corpora are always flanked by acoustic silences, with the majority of coughing events occurring at interlocutor turn exchanges. Assuming that urges to cough can arise at anytime, we interpret the restriction of cough occurrence to pauses and turns as evidence that talkers can and do delay coughing until the end of their intonational phrases or to floor exchanges with their interlocutor. This further indicates a degree of linkage between the non-speech motor system and the prosodic speech planning system. [Work supported by NIH.]

4aSC17. Tongue bracing under bite block perturbation. Monika M. Luszczuk (Inst. of Speech Therapy and Appl. Linguist, Maria Curie Sklodowska Univ. (UMCS), Lublin, Poland, ul. Sowińskiego 17, Lublin 20-040, Poland, monika.luszczuk@gmail.com), Murray Schellenberg (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), Yadong Liu (Dept. of Linguist, Univ. of Br. Columbia, Hong Kong, Hong Kong), and Bryan Gick (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Speech sounds have been shown to adapt quickly but imperfectly under bite block perturbation, supporting opposing acoustic vs. articulatory compensation mechanisms [Gay *et al. JASA* 69: 802. 1981; Flege *et al. JASA* 83: 212. 1998]. The present study considers whether lingual bracing may provide insight into these apparently conflicting findings. Tongue bracing against the teeth or palate is a pervasive posture maintained during normal speech [Gick *et al. JSLHR*. 60:494. 2017]; we aim to test whether the tongue adapts its bracing position rather than adapting each speech movement individually, providing a single, postural parametric mechanism for responding to jaw perturbation. Results of an experiment will be presented in which native English-speaking participants read aloud passages normally and under bite block conditions translating the jaw in forward, backward or lateral directions, and to varying degrees of opening. Coronal ultrasound imaging results will be reported, measuring positions of the lateral tongue for indications of stable bracing postures. Implications of these findings will be discussed for models of speech production. [Funding from NSERC.]

4aSC18. Lateral bias in lingual bracing during speech. Bryan Gick, Megan Keough, Oksana Tkachman (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca), and Yadong Liu (Linguist, Univ. of Br. Columbia, Hong Kong, Hong Kong)

Human bodies exhibit lateral biases between many laterally symmetrical body parts (e.g., hands, feet, eyes, and ears) that increase the efficiency of behaviour and functionality of a system. We report on lateral biases observed in the tongue during speech. The tongue is bilaterally braced against the back teeth and hard palate throughout speech, but is interrupted for the production of some laterals and occasional low vowels; some evidence suggests the movement away from the braced posture may be produced by lowering one side of the tongue first and that the leading side is consistent within speaker [Gick *et al.* 2017, JSLHR 60(3), 494]. We report findings on lateral bias in English speakers, on its correlation with other lateral biases of the speakers, and what this may imply about the origins of this bias. Preliminary results indicate some variation, with a population-level bias (preference for one side over the other), suggesting that the bias may develop with cortical modulation in much the same way that handedness is thought to arise. [Funding through NSERC.]

4aSC19. Cross-linguistic lateral bracing: An ultrasound study. Felicia Tong (Linguist, Univ. of Br. Columbia, 6368 Stores Rd., Rm. 4, Vancouver, BC V6T 1Z2, Canada, feliciatong@yahoo.com.hk), Yadong Liu (Linguist, Univ. of Br. Columbia, Hong Kong, Hong Kong), Dawoon Choi (Psych., Univ. of Br. Columbia, Vancouver, BC, Canada), Megan Keough, and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Lateral bracing refers to intentional stabilizing of tongue contact with the roof of the mouth along the upper molars or the hard palate. Previous research has found evidence of lateral bracing in individual speakers of six languages [Cheng *et al.* 2017. *Can. Acoust.* 45, 186]. The current study examines lateral bracing cross-linguistically at a larger scale using ultrasound technology to image tongue movement. We tracked and measured the magnitude of vertical tongue movement at three positions (left, right, and middle) in the coronal plane over time using Flow Analyzer [Barbosa, 2014. *J. Acoust. Soc. Am.* **136**, 2105] for optical flow analysis. Preliminary results across all languages (Cantonese, English, French, Korean, Mandarin, and Spanish) show that the sides of the tongue are more stable than the center and maintain a relatively high position in the mouth throughout speech. The magnitude of movement at the sides is significantly smaller than at the center of the tongue. Further, lateral releases vary in frequency for different languages. This evidence supports the view that bracing is a physiological property of speech production that occurs irrespective of the language spoken. [Funding from NSERC.]

4aSC20. Coarticulation of speech and smile movements. Terrina Chan, Ryan C. Taylor, Esther Y. Wong, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, terrinachan@gmail.com)

Facial expressions and speech movements can impose conflicting demands on articulators. For example, the lip spreading movement associated with smiling is incompatible with bilabial closures for /m/, /b/ or /p/. Anecdotal evidence suggests this conflict may resolve as labiodental stop variants (see http://phonetic-blog.blogspot.ca/2012/03/u.html), though this discussion has been controversial [Ladefoged & Maddieson 1996, p. 18]. The simplest model of coarticulation—one of unmediated superposition of muscle activations [Gick *et al.*, 2013, *POMA* 060207]—predicts that the outcome of this conflict should be determined by summing opposing forces due to competing muscle activations. If so, varying degrees of smile and varying degrees of closure force (e.g., for different stop consonants) should be expected to produce distinct outputs. Previous work suggests that closures for /m/, /b/ and /p/ vary (increasingly) in both intraoral pressure [Lubker & Parris 1970, *JASA* 47: 625] and muscle force [Gick *et al.* 2012, *JASA* 131: 3345]. An experiment will be presented in which bilabial stops are produced under varying smile conditions. Preliminary results indicate that labiodental stop variants occur more frequently for lower-force stops under higher-force smile conditions, as predicted. Implications for models of coarticulation will be discussed. [Funding from NSERC.]

4aSC21. A computational study of the role of laryngeal muscles in vocal fold posturing. Xudong Zheng, Ngoc Pham, Biao Geng, and Qian Xue (Mech. Eng., Univ. of Maine, 213 Boardman Hall, Orono, ME 04469, xudong.zheng@maine.edu)

This work aims at using a three-dimensional finite element method to simulate the contraction of different laryngeal muscles, and to study the role of each or groups of muscles in vocal fold posturing. A Hill-based contractile model is coupled in the finite element analysis to capture the active response of the vocal folds, and a fiber-reinforced model is employed to model the passive response. A 1D flow model and 1D acoustic solver are coupled with the vocal fold model to simulate the flow-structure-acoustics interaction. In the simulation, the geometry of the vocal folds and cartilages is reconstructed from a MRI scan. The coupled Hill-based contractile model and fiber-reinforced tissue model shows good agreement with literature data for simulating dynamics, concurrent tissue stimulation and stretching. It is found that the contraction of cricothyroid (CT) muscle alone lengthens the vocal folds by rotating the thyroid cartilage clockwise, around the cricoid-thyroid joint, whereas the contraction of cricoarytenoid (CA) muscle causes vocal fold abduction. The contraction of TA muscle and arytenoid muscles, on the other hand, adduct the vocal folds. The effect of muscle contraction on subglottic pressure during voice production is also shown in flow-structure-acoustics interaction simulations.

4aSC22. Effects of tongue elevation speed and glottal flow control on the production of sibilant /s/. Tsukasa Yoshinaga (Graduate School of Eng. Sci., Osaka Univ., 1-3 Machikaneyama, Toyonaka, Osaka 560-8531, Japan, t.yoshinaga@me.es.osaka-u.ac.jp), Kazunori Nozaki (Osaka Univ. Dental Hospital, Suita, Japan), and Shigeo Wada (Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Japan)

To clarify the effects of tongue elevation speed and glottal flow control on the production of /s/ in VCV sequence, the experimental measurements were conducted using a simplified vocal tract model. The simplified model was constructed based on five cross-sectional shapes of the vocal tract of a subject pronouncing /s/, which were measured with CT scans. The tongue elevation speed and volume flow rate at the glottal inlet of the model were controlled by using a stepper motor and an electromagnetic proportional control valve, respectively. The flow velocity at the gap between front teeth and the sound propagating from the model were measured by a hot-wire anemometer and a microphone simultaneously. By controlling the tongue elevation speed and the flow rate of the simplified model, the spectrogram of /s/ in Japanese word /usui/ was reproduced. The result of the flow measurement showed that the velocity fluctuation and the sound propagation started almost at the same time when the spectrogram was reproduced. In contrast, by lowering the tongue elevation speed, the velocity fluctuation preceded appearance of the sound generation. This indicates that the tongue elevation speed has a significant effect on controlling the sound source generation of /s/ in word pronunciation.

4aSC23. Using ultrasound to examine contrastive hyperarticulation. Kathleen C. Hall, Geoff Fullerton (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, kathleen.hall@ubc.ca), and Kevin McMullin (Dept. of Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

The current paper examines the extent to which there is cross-linguistic evidence for the hyperarticulation of more phonologically contrastive vowel sounds. Hall *et al.* [2017, *JCAA* 45: 15] found that tense vowels in English are produced with more total tongue movement when in positions where they are more phonologically contrastive. This finding was established through the use of Optical Flow Analysis [Horn & Schunck 1981] on ultrasound videos of the tongue. The “more contrastive” positions were those in which the vowels could contrast with their lax vowel counterparts, while the “less contrastive” positions were those in which no such contrast was possible. While there was evidence that the degree of contrast affected the tongue movements, the data were somewhat confounded by the fact that the more contrastive positions were largely closed syllables, while the less contrastive positions were largely open syllables. In the current paper, we first replicate the original results with more tightly controlled phonetic contexts in English and then examine analogous results for Canadian French. Crucially, in Canadian French, [e] vs. [ɛ] contrast in open syllables and not in closed, such that the effects of syllable position and phonological contrast can be teased apart. [Funded by SSHRC.]

4aSC24. Registration and fusion of 3D head-neck MRI and 3D/4D tongue ultrasound. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Static 3D head-neck MRI can provide excellent coverage of the entire vocal tract and surrounding anatomy with good spatial resolution and signal-to-noise ratios. Real-time MRI takes advantage of these strengths and adds temporal resolution for dynamic speech production, but depends on regular repetition of the same speech utterances for several minutes. (Dynamic or cine MRI also depends on regular repetitions, but the process is stroboscopic rather than reconstructive, and the end result is a reduced number of time frames with longer frame durations.) Real-time MRI is not suitable for non-repetitive speech utterances. In contrast, 3D/4D ultrasound can provide excellent spatial resolution with good temporal resolution of non-repetitive speech utterances, but with lower signal-to-noise ratios and poor anatomical coverage. This study presents methods for registering and fusing static head-neck MRI and 3D/4D tongue ultrasound data, taking advantage of the strengths of both MRI and 3D/4D ultrasound to investigate the production of non-repetitive speech utterances.

4aSC25. Towards higher precision in vocal tract length estimation. Stefon M. Flego (Linguist, Indiana Univ., Bloomington, 720 S College Mall Rd. Apt. N5, Bloomington, IN 47401, sflego@indiana.edu)

Approaches to vocal tract length (VTL) estimation typically differ in how much credibility is allocated to certain formants as predictors of length, but are alike in allocating equal credibility to all vowel spectra. The latter may be problematic, as asymmetries in the phonotactic frequency of certain vowels and measurement errors for higher formants can influence VTL estimates. Herein, an additional parameter is proposed which privileges vowel spectra that approximate a uniform tube. The proposed metric for this proximity is standard variance (σ) in Φ, where $\Phi{=}F_n/(2n\text{-}1)$, n represents the integer label of the formant, and σ_Φ for a uniform tube is 0. The five estimators detailed in Lammert & Narayanan (2015) were tested on the speech of two adult males. Each of the estimators were run using all vowel spectra as well as only those for which $\sigma_\Phi < 50$ Hz. Estimates from all vowel spectra had σ_{VTL} well above 1 cm, which is undesirably large, given that the adult vocal tract only ranges from 13 to 20 cm. However, estimates from vowel spectra with $\sigma_\Phi < 50$Hz had σ_{VTL} less than 1 cm. This higher precision is especially striking since spectra with $\sigma_\Phi < 50$ Hz constituted a small fraction of the total.

4aSC26. Individual differences and their effect on the nature of speech motor errors in younger and older adults. Katherine M. Dawson (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gradcenter.cuny.edu), Mark Tiede (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York Graduate Ctr., New York, NY)

Older adults have been reported to have a greater propensity towards speech errors [MacKay & James, *Psych & Aging*, *19*(1), 93–107 (2004)],but as with many age-related effects, there is considerable variability in results and a need to identify behavioral predictors. The present study employs a motor speech paradigm, whereby both younger and older adults repeated word pairs (e.g., "top-cop") timed to a metronome in an accelerating rate task. The metronome had a steady rate for half the task (8s) and then increased at a linear rate. Participants also completed evaluations of motor speech, sensory, and cognitive abilities, so that these individual characteristics can be correlated with speech error tendencies. Articulatory data of speech movements were collected in the form of ultrasound tongue contours and lip aperture derived from facial video. Articulatory results are discussed in terms of speech error gradience in the context of previous work [Pouplier, *Proc. XVth ICPhS,* 2245–2248 (2003)], with emphasis on how well speakers adapt to the changing metronome rate, and how this affects error incidence in younger versus older adults.

4aSC27. Reduced coarticulation and aging. Cécile Fougeron, Daria D'Alessandro, and Leonardo Lancia (Laboratoire de Phonétique et Phonologie, CNRS-U. Sorbonne Nouvelle, 19 rue des Bernardins, Paris 75005, France, cecile.fougeron@univ-paris3.fr)

Lifespan changes in speech have mostly been documented with respects to children's development, while little is known about its evolution throughout adulthood. More particularly our knowledge on the effect of aging on speech and voice is sparse. Changes have been found in voice quality [e.g., Xue & Hao, 2003], the speech rate has been described to slow down [e.g., Staiger *et al.* 2017), and pitch is said to raise for older males and to lower for older females [e.g., Harnsberger *et al*., 2008]. The present study aims to investigate the effect of aging on $\text{Vowel}_{\text{to\_Vowel}}$ anticipatory coarticulation in French. This effect is tested according to vowel duration and to differences in regional varieties. Data from 240 speakers (half female) distributed across three age groups (20–45), (50–69), and (70–80) have been extracted from the $\text{MonPaGe}_{\text{HA}}$ database [Fougeron *et al.* 2018] which includes speakers from four regional varieties: France, Belgium, Switzerland, and Quebec. The influence of V2 (/a/ or /i/) on V1 (/a/) is measured as a lowering of F1 and a rise of F2. Results show that coarticulation and vowel duration varies with regional variety and that vowels of older speakers are lengthened. More interestingly, results show a reduction of coarticulation with age.

4aSC28. Improving the quality of speech in the conditions of noise and interference. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org) and Krzysztof Kąkol (PGS Software SA, Gdansk, Poland)

The aim of the work is to present a method of intelligent modification of the speech signal with speech features expressed in noise, based on the Lombard effect. The recordings utilized sets of words and sentences as well as disturbing signals, i.e., pink noise and the so-called babble speech. Noise signal, calibrated to various levels at the speaker's ears, was played over two loudspeakers located 2 m away from the speaker. In addition, the recording session included utterances in quiet, which constitute a reference to the received speech signal analysis with the Lombard effect. As a part of the analysis, the following parameters were examined with regard to prosody: fundamental frequency F0, formant frequencies of F1 and F2, duration of the utterance, sound intensity, etc., taking into account individual sentences, words, and vowels. The PRAAT program was used to process and analyze speech signals. Next, a method for modifying speech with the features of speech spoken in noise was proposed. Subsequent analyzes have shown that noisy speech modified by the Lombard effect features is characterized by higher values of the PESQ (perceptual evaluation of speech quality) speech quality indicator compared to noisy speech without the features incorporated.

4aSC29. How much formant variability in linear prediction-based measurements can be explained by F0 variability? Wei-Rong Chen, D. H. Whalen, and Christine H. Shadle (Haskins Labs., 300 George St. STE900, New Haven, CT 06511, chenw@haskins.yale.edu)

Previous studies have used speech variability as a measure of speech development; for instance, children reduce the variability in their formant frequencies as they grow, indicating increases in speech motor control. However, formant measurements in most of these studies are computed using variants of linear prediction coding (LPC), which is known to exhibit a bias towards the nearest harmonic of F0, especially at high F0 [Shadle *et al*. (2016). JASA. 139(2), 713–727]. The proportion of the reported formant variabilities that can be attributed to changes in F0 is still unknown; thus, the true variability in formant frequency is also unknown. Therefore, we used published values of formant and F0 variabilities for /a/ produced by children (age range: 4~19 + yrs), and, for each age group, we synthesized 1000 vowels with the same formants (F1~F3) fixed at the reported group means, but randomly varied F0 using a normal distribution (μ = F0 group mean, σ = F0 standard deviation (SD)). We then measured formants in those synthesized vowels using LPC; the SDs of the measured formants were considered to be the F0-biased formant variability (true formant SD = 0). A ratio of F0-biased SD to reported formant SD for each age group indicates how much of the reported formant variability can be explained by F0 changing in children's speech. The results show, on average, that the purported formant variabilities in previous studies that can be explained by F0 bias are 82.2 % for F1, 31.1% for F2, and 17.4% for F3.

4aSC30. Pitch duration as a cue for declination. Lihan Wu (School of Foreign Lang. Studies, Guangxi Univ. for Nationalities, Nanning, Guangxi 530006, China, linhuadance@hotmail.com) and Hua Lin (Linguist, Univ. of Victoria, Victoria, BC, Canada)

An utterance of a language often demonstrates the effect of declination, usually understood broadly as a reduction in certain physical or acoustic signals. Previous research on declination focuses primarily on the acoustic measures of pitch (such as the movement of pitch or the alternation of pitch span) or intensity. Neglected in the matter is the third acoustic dimension of speech duration. This paper reports on an experiment on declination focusing on *pitch duration*. The language studied is Mandarin Chinese. Six native Mandarin speakers are recruited. A total of 864 utterances of 2-to-9 syllables in four tones and four functional intonations are recorded and analyzed on Praat. The results show that the declination of pitch duration goes side by side along the pitch declination, both of which share the same physiological basis. Specifically, (1) the average pitch duration within the prosodic unit decreases gradually from the sentence-initial prosodic unit to the sentence-final one, (2) the pitch duration of the left-most syllable of the prosodic unit decreases gradually from the sentence-initial prosodic unit to the sentence-final one, and (3) the pitch duration of the right-most syllable of the prosodic unit decreases slightly from the sentence-initial prosodic unit to the sentence-final one.

4aSC31. Fully-automated tongue detection in ultrasound images. Elham Karimi (Dept. of Elec. Eng., École de technologie supérieure, 1100 Notre Dame Ouest, Bureau A-2464, Montreal, QC H3C 1K3, Canada, elham.karimi.1@etsmtl.net), Lucie Menard (Dept. of Linguist, Université du Québec à Montréal, Montreal, QC, Canada), and Catherine Laporte (Dept. of Elec. Eng., École de technologie supérieure, Montreal, QC, Canada)

Tracking the tongue in ultrasound images provides information about its shape and kinematics during speech. Current methods for detecting/tracking the tongue require manual initialization or training using large amounts of labeled images. This work introduces a new method for extracting tongue contours in ultrasound images that requires no training or manual intervention. The method consists in: (1) application of a phase symmetry filter to highlight regions possibly containing the tongue contour; (2) adaptive thresholding and rank ordering of grayscale intensities to select regions that include or are near the tongue contour; (3) skeletonization of these regions to extract a curve close to the tongue contour; and (4) initialization of an accurate active contour from this curve. Two quality measures were also developed that predict the reliability of the method so that optimal frames can be chosen to confidently initialize fully automated tongue tracking. Experiments were run on 16 free speech ultrasound recordings from healthy subjects and subjects with articulatory impairments due to Steinert's disease. Fully automated and semi automated methods result in mean sum of distances errors of 0.92mm +/- 0.46 mm and 1.04 mm +/ 0.5513 mm, respectively, showing that the proposed automatic initialization does not significantly alter accuracy.

4aSC32. Speech adaptation to palatal perturbation: Evidence for sensorimotor reorganization across the workspace. Douglas Shiller, Guillaume Barbier (School of Speech-Lang. Pathol. & Audiol., Université de Montréal, PO Box 6128, succursale Centre-ville, Montreal, QC H3C 3J7, Canada, douglas.shiller@umontreal.ca), Lucie Menard (Linguist, Université du Québec à Montréal, Montreal, QC, Canada), and Shari Baum (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada)

Competent speakers demonstrate a high degree of precision in speech production, but also considerable flexibility in speech motor patterns used to attain speech goals. This flexibility is evident in studies examining adaptation to sensory perturbations, including physical manipulations that alter both auditory and somatosensory feedback (e.g., using a palatal prosthesis to disrupt the production of the alveolar fricative /s/). While the acoustic effects of such perturbations (and subsequent adaptation) have been well explored, their underlying articulatory basis remains poorly understood. In this study, we explored speech adaptation to a palatal perturbation in 9 adult speakers using acoustic and articulatory measures of adaptation (using electromagnetic articulography) during the production of various consonants and vowels. Acoustic measures showed effects of both the perturbation and subsequent compensation (following 15 minutes of speech practice) across the consonants and vowels. Analysis of tongue kinematics confirmed the global nature of these effects, including significant changes in sounds involving no physical interaction with the palate. Furthermore, directional analyses revealed a degree of fine-tuning in motor adaptation across phonemes. The findings indicate that even localized changes in palate shape induce complex compensatory changes in speech motor control across the articulatory workspace.

4aSC33. Cepstral coefficients successfully distinguish the front Greek fricatives. Laura Spinu (Communications & Performing Arts, CUNY - Kingsborough Community College, 2001 Oriental Boulevard, Brooklyn, NY 11235-2398, Laura.Spinu@kbcc.cuny.edu), Jason Lilley (Ctr. for Pediatric Auditory & Speech Sci., Nemours Biomedical Res., Wilmington, DE), and Angeliki Athanasopoulou (School of Lang., Linguist, Literatures, and Cultures, Univ. of Calgary, Calgary, AB, Canada)

In the current study, we explore the factors underlying the well-known difficulty in acoustic classification of front fricatives (McMurray & Jongman, 2011; Maniwa *et al*., 2009) by taking a closer look at the production of 29 native Greek speakers. Our corpus consists of Greek fricatives from five places of articulation and two voicing values [f, v, *θ*, ð, s, z, ç, ʝ, x, ɣ] produced in nonce disyllabic words before [a, o, u] in stressed syllables (e.g., [ˈ*θ*akos]). We apply a relatively novel classification method based on cepstral coefficients, previously successful with obstruent bursts (Bunnell *et al*., 2004), vowels (Ferragne & Pellegrino, 2010), and Romanian fricatives (Spinu & Lilley, 2016). Our method yields the best correct classification rates reported to date for front fricatives: Present study: 88%; English: 66% (Jongman *et al.,* 2000), Greek: 55.1% (Nirgianaki, 2014). The important cues for the successful classification are the vowel following the target fricative and the second region of frication noise. Our study adds to the body of work aimed at identifying techniques for quantifying and categorizing large samples of speech. Obtaining higher classification rates than before takes us one step closer to understanding the properties of "difficult" sounds like the front fricatives.

4aSC34. Nasal rustle: An evidence-based description, II. 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), Srujana Padakanti, Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH), Ann W. Kummer (Speech-Lang. Pathol., Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), and Suzanne E. Boyce (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

This research extends a previous study investigating the sound source of audible nasal emissions associated with a small velopharyngeal (VP) opening (herein "nasal rustle," but also "nasal turbulence" elsewhere). This extension includes more speech sounds than the original study, which only considered /s/. Speech sounds susceptible to nasal rustle were recorded from ten pediatric patients with VP insufficiency via high-speed nasopharyngoscopic video at the superior VP port and simultaneous nasometry. The total pixel intensity was summed in each video frame to generate a video intensity signal. For each speech sound token, a cross-correlation was computed between this video intensity signal and each of the nasometer's acoustic signals; this measures similarity in frequency content between motion at the superior VP port and acoustics exterior to the lips and nares, respectively. The difference between the video-nasal cross-correlation and the video-oral cross-correlation over a frequency bandwidth including the dominant frequencies observed in the video intensity signal (<100 Hz) has a positive association with the number of mucus movements, meaning the motion of mucous secretions correlates more strongly with acoustic energy exterior to the nares than exterior to the mouth. This indicates that mucus motion could be a dominant sound source of nasal rustle.

4aSC35. Contribution of the tongue tip retraction in the articulation of high vowels. Hayeun Jang (Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, hayeunja@usc.edu)

This study shows that the retraction of the tongue tip contributes to the raising of the tongue body in the articulation of high vowels /i/ and /u/ by using qualitative simulations of tongue deformation by using a biomechanical 3D tongue model of Artisynth (Lloyd *et al*. 2012) and the analysis of tongue configuration in X-ray microbeam data (Westbury 1994). The simulations qualitatively replicated the tongue shapes at the temporal mid-point of /i/ and /u/ in rtMRI data by manipulating the activation values of tongue muscles in Artisynth. The results show that the tongue tip retraction helps to raise the tongue body higher in both /i/ and /u/. In the simulations of /i/, the tongue tip retraction raises the tongue body even without moving the tongue body. The results of x-ray microbeam analysis confirm those findings of simulations. The mixed linear regression model of tongue configurations in the tasks of sentence and paragraph production shows that more tongue tip retraction (a shorter horizontal distance between two anterior tongue pellets) significantly correlates with a higher position of the tongue body (higher one between two posterior tongue pellets) in both /i/ and /u/ and the correlation is stronger in /i/ than in /u/.

4aSC36. Limitations of the source-filter coupling in phonation. Debasish R. Mohapatra and Sid Fels (Dept. of Elec. and Comput. Eng., Univ. of Br. Columbia, 2366 Main Mall, Vancouver, BC V6T 1Z4, Canada, d.mohapatra@alumni.ubc.ca)

The coupling of vocal fold (source) and vocal tract (filter) is one of the most critical factors in source-filter articulation theory. The traditional linear source-filter theory has been challenged by current research which clearly shows the impact of acoustic loading on the dynamic behavior of the vocal fold vibration as well as the variations in the glottal flow pulses' shape. This paper outlines the underlying mechanism of source-filter interactions; demonstrates the design and working principles of coupling for the various existing vocal cord and vocal tract biomechanical models. For our study, we have considered self-oscillating lumped-element models of the acoustic source and computational models of the vocal tract as articulators. To

understand the limitations of source-filter interactions which are associated with each of those models, we compare them concerning their mechanical design, acoustic and physiological characteristics and aerodynamic simulation. References: 1. Flanagan, J. L. (**1968**) "Source-system interaction in the vocal tract," Ann. N.Y. Acad. Sci. **155**, 9–17. 2. Lieberman, P., and Blumstein, S.E. (**1988**) Speech physiology, speech perception, and acoustic phonetics (Cambridge University Press, Cambridge, Mass). 3. Titze, I. R. (**2008**) "Nonlinear source-filter coupling in phonation: Theory," *J. Acoust. Soc. Am.* **123,** 2733–2749.

4aSC37. Articulatory feature extraction from ultrasound images using pretrained convolutional neural networks. Kele Xu (School of Comput., National Univ. of Defense Technol., 16 Rue Flatters, Paris 75005, France, kelele.xu@gmail.com) and Jian Zhu (Dept. of Linguist, Univ. of Michigan, Ann Arbor, MI)

Feature extraction is of great importance to ultrasound tongue image analysis. Inspired by the recent success of deep learning, we explore a novel approach to feature extraction from ultrasound tongue images using pretrained convolutional neural networks (CNN). The bottleneck features from different pre-trained CNNs, including VGGNet and ResNet, are used as representations of the ultrasound tongue images. Then an image classification task is conducted to assess the effectiveness of CNN-based features. Our dataset consists of 20,000 ultrasound tongue images collected from a female speaker of Mandarin Chinese, which were manually labeled as containing one of the following consonants: /p, t, k, l/. Experiment results show that the Gradient Boost Machines (GBM) classifiers trained on the CNN-based features achieve the best performance, with a classification accuracy of 92.4% for ResNet and 91.6% for VGGNet, outperforming the benchmark GBM classifier trained on the features extracted using Principal Component Analysis (PCA), which only achieves an accuracy of 87.5%. In this preliminary dataset, our method of feature extraction is found to be superior to the PCA-based method. This work demonstrates the potential of applying the pre-trained convolutional neural networks to ultrasound tongue image analysis task.

4aSC38. Hand gesture density in prosodically important speech regions. Samantha G. Danner (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, sfgordon@usc.edu)

Hand movements during speech are systematically linked with a variety of "landmarks" in the acoustic speech signal, including prosodically important regions in speech. While many researchers have studied qualitative aspects of how hand movement coordinates with speech, there are few studies that consider quantitative properties of manual gestures during speech and their coordination with speech prosody. We use optical flow to automatically detect manual gesture and identify velocity peaks in the right hand movement signal. Hand gestures for six speakers are z-scored and classified into three "sizes" as a function of velocity peak magnitude. We measure whether manual gesture density—i.e., the occurrence frequency of velocity peaks—differs among regions around prosodic landmarks in the accompanying acoustic speech signal, including at speech turns, phrase boundaries, acoustic amplitude prominences, and regions containing both a pitch accent and an amplitude prominence, with respect to a baseline of all other regions of speech. Results show that manual gestures of all sizes are more frequent at phrase boundaries and at acoustically and prosodically prominent regions of speech than at turn boundaries and non-prominent speech regions. This suggests that gesture density may be a communicative component highlighting prosodically important regions of speech. [Work supported by NIH.]

4aSC39. Articulatory and acoustic investigations into gestures of Mandarin retroflex fricatives . Yung-hsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., Zhongxiao E. Rd., Sec. 3, No. 1, Taipei 106, Taiwan, shawnchang@mail.ntut.edu.tw)

This study investigates the inventory and acoustic realizations of articulatory gestures of the Mandarin retroflex (or arguably, post-alveolar) fricative /ʂ/. Tongue shapes were obtained with ultrasound images and categorized as retroflexed or bunched, in light of the literature on the North American English rhotic /ɹ/ production. Velarization, a secondary articulatory gesture of retroflex consonants, was also coded in the ultrasound imaging analysis. Lip rounding, found to be an optional enhancing gesture in Mandarin retroflexion, was identified with lip video data. The acoustic measurements included the spectral center of gravity obtained from the middle of the frication and F2 frequency measured at the onset of the vowel following the fricative. The analyses showed that the spectral center of gravity only varied as a function of the vowel context. F2 at vowel onset was affected by vowel context, velarization and the interaction of vowel context and lip rounding. The preliminary results failed to support previous speculations that the two tongue shapes in Mandarin retroflexes result in different acoustics. We consider including more acoustic measures to further explore the acoustic consequences of articulatory variations in Mandarin retroflex fricatives.

4aSC40. A mechanical deformation model of the tongue during speech production considering personalized anatomical structure. Shun Takenaka, Tsukasa Yoshinaga (Graduate School of Eng. Sci., Osaka Univ., 1-3, Toyonaka, Osaka 560-8531, Japan, s.takenaka@me.es.osaka-u.ac.jp), Kazunori Nozaki (Dental Hospital, Osaka Univ., Suita, Japan), Satoshi Ii (Graduate School of Systems Design, Tokyo Metropolitan Univ., Hachioji, Japan), and Shigeo Wada (Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Japan)

In the previous studies, several tongue models have been developed to investigate the relationship between the contraction of tongue muscles and the deformation of the tongue during speech. However, the tongue deformation is highly subject-specific and their modellings are insufficient to uncover the detailed mechanism. In this study, a mechanical tongue model considering personalized anatomical structure of tongue myofiber is proposed. A set of reference geometries of tongue and jaw is constructed from the medical images of computer tomography (CT) and the magnetic resonance imaging (MRI) for a Japanese male adult. In addition, we reflect the information of tongue myofiber orientations evaluated from the medical images of diffusion tensor imaging (DTI) on the model using a smoothing extrapolation technique. The tongue deformation is calculated by solving a force-equilibrium equation using a continuous Galerkin finite element method with a hyperelastic material, where the deformation is driven by active contraction of the tongue muscles. We examined the tongue deformations under several contraction conditions during speech and confirmed reasonable numerical reproductions of actual tongue motion and shape. These results indicate that the proposed model reflecting personalized information including myofiber orientations has a potential to identify the tongue muscle contractions in speech production.

4aSC41. SOUND STREAM: Towards vocal sound synthesis via dual-handed simultaneous control of articulatory parameters. Pramit Saha, Debasish R. Mohapatra, Venkata Praneeth Srungarapu, and Sid Fels (Dept. of Elec. and Comput. Eng., Univ. of Br. Columbia, 2366 Main Mall, Vancouver, BC V6T 1Z4, Canada, pramit@ece.ubc.ca)

This paper introduces Sound stream: a low-cost, tangible and ambidextrous controller which drives a dynamic muscle-based model of the human vocal tract for articulatory speech synthesis. The controller facilitates the multidimensional inputs which are mapped to the tongue muscles in a biomechanical modeling toolkit Artisynth using a microcontroller. As the vocal tract is a complex biological structure containing many muscles, it is a challenging and computationally expensive task to accommodate control for every muscle in the proposed scheme. So, we have followed a simplified approach by controlling the selective muscles for the efficient articulatory speech synthesis. The goal for designing an ambidextrous controller is to create new possibilities of controlling multiple parameters to vary the tongue position and shape simultaneously for generating various expressive vocal sounds. As a demonstration, the user learns to interact and control a mid-sagittal view of the tongue structure in Artisynth through a set of sensors using both hands. The Sound-Stream explores and evaluates the appropriate input and mapping methods to design a controllable speech synthesis engine. 1. Wang, J. *et al.* (**2011**) "Squeezy: Extending a multi-touch screen with force sensing objects for controlling articulatory synthesis," in Proceedings on New Interfaces for Musical Expression, Oslo, Norway, pp. 531–532.

4aSC42. Co-mingling effects of dialectal and stylistic variation due to choice of speaker normalization. Wil A. Rankinen (Commun. Sci. and Disord., Grand Valley State Univ., 515 Michigan St. NE, Ste. 300, Office 309, Grand Rapids, MI 49503, wil.rankinen@gvsu.edu) and Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN)

This paper explores how a speaker normalization routine can result in co-mingling effects of dialectal and stylistic variation. To this end, the paper examines a database of vowel formants in a passage and a wordlist task collected from 88 monolingual American English speakers in Michigan's Upper Peninsula; this corpus is further stratified by ethnic-heritage, age, sex and educational attainment. Two different speaker normalization routines, Lobanov (Adank *et al*., 2004) and Labov ANAE (Labov *et al*., 2006), were used; however, each routine yielded differences in a paralleled analysis. The Lobanov normalization routine removes much of the ethnic-heritage effect that appears in the analysis using the Labov ANAE normalization routine. Examining the cause of these different results reveals that 1) the Lobanov routine removes any mean shifts in formant values that run orthogonal to anatomical variation (i.e., vocal tract length effects) and 2) the ANAE routine only removes shifts which are correlated across formants. The ethnic-heritage effect in this dataset involves mean shifts that are negatively correlated to typical anatomical variation, which results in the ANAE routine potentially amplifying such effects. We discuss the import of these findings for vowel space variation and language change.

4aSC43. Reconstruction of the glottal pulse using a subband technique on kazoo recordings. Alexandre M. Lucena, Mario Minami (CECS, Federal Univ. of ABC, Av. dos Estados, 5001, Santo André, São Paulo 09210-580, Brazil, ale.mlucena@gmail.com), and Miguel A. Ramirez (Electron. Systems, Univ. of São Paulo, São Paulo, São Paulo, Brazil)

The kazoo, a wind instrument, generates its typical sound when stimulated by voiced speech. Using this instrument, this paper proposes a novel technique to recover the glottal pulse excitation of its player. We applied multiband frequency techniques to the kazoo signal to compare the results with those obtained from the corresponding recordings of an electroglottograph (EGG). With the player's management over his embouchure on the instrument, one can make recordings for spoken and singing speech as well as recitative, at the instrument's resonator cap, which closely fit the EGG recordings. After a spectrogram analysis, it was possible to detect in the lower frequency band of the kazoo signals, the spectral envelope and, in the higher frequency band, the pitch harmonics mixed with the spectral decay of the glottal pulse. A quadrature mirror filter (QMF) was designed, providing this source-filter separation. Additionally, a reverse spectral band replication (SBR) technique was applied, which consists in recovering the lower frequency band by the demodulation of the higher frequency band followed by a total energy spectral gain adjustment, where a new signal was generated and then evaluated. At the end, a subjective evaluation, SNR, and SD measures prove the efficiency of the proposed method.

4aSC44. Toward the automatic detection of manually labeled irregular pitch periods. Olivia Murton (Speech and Hearing BioSci. and Technol., Harvard Med. School, One Bowdoin Square, 11th Fl., Boston, MA 02114, omurton@g.harvard.edu), Sophie Rosas-Smith (Comput. Sci., Wellesley College, Wellesley, MA), Jeung-Yoon Choi (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA), Daryush Mehta (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

Irregular pitch periods (IPPs) occur in a wide variety of speech contexts and can support automatic speech recognition (ASR) systems by signaling word boundaries, phrase endings, and certain prosodic contours. IPPs can also provide information about emotional content, dialect, and speaker identity. The ability to automatically detect IPPs is particularly useful because accurately identifying IPPs by hand is time-consuming and requires expertise. In this project, we use an algorithm developed for creaky voice analysis by Kane *et al*. (2013) incorporating features from Ishi *et al*. (2008) to automatically identify IPPs in recordings of speech from the American English Map Task (AEMT) database. Short-term power, intra-frame periodicity, inter-pulse similarity, subharmonic energy, and glottal pulse peakiness measures are input into an artificial neural network to generate frame-by-frame creak probabilities. To determine a perceptually relevant threshold probability, the creak probabilities are compared to IPPs hand-labeled by experienced raters. Preliminary results from four AEMT conversations yielded an area under the receiver operating characteristic curve of 0.898 on average. A creak probability threshold of 0.0083 yielded equivalent sensitivity and specificity of 81.1%. This study indicates generally good automatic detection of hand-labeled IPPs and explores the effects of linguistic and prosodic context.

4aSC45. Belching as a non-lexical speech object: Evidence from pop-culture. Brooke L. Kidner (Linguist, Univ. of Southern California, 620 McCarthy Way, #273, Los Angeles, CA 90089, bkidner@usc.edu)

Belching is normally considered an involuntary speech-tract vocalization. When speaking, involuntary speech-tract gestures such as belching often compete with speech for articulation by our vocal apparatus. There are non-lexical paralinguistic items (i.e., "ugh") that also convey speech meaning, but traditionally belching is not regarded as a speech object. This study presents a case where belching appears to be an intentional speech act, occurring with clearly defined parameters and not in competition with speech. Data comes from the character Rick from TV's Rick and Morty. Transcripts were collected and compared with the audio data. 105 belches were found in the 8 episodes investigated: all belches, except 1, were not written into the original scripts or transcriptions. Acoustic analysis showed no significant disruptions between belching and speech, supporting the conclusion that belches do not compete with speech. The most frequent occurrence of belching is word-medial, after the initial segment, which is a common infixation pattern cross-linguistically. Additional acoustic analysis showed belching occurred regularly before a stressed syllable, an established pattern for infixation in English. This pattern provides evidence that these belches are behaving phonologically as para-phonemic items. The results show that belching can be intentional and behave like paralinguistic items in human language and communication.

4aSC46. Accidental gaps in Mandarin Chinese tones. Shao-Jie Jin and Yu-an Lu (Foreign Lang. and Literatures, National Chiao Tung Univ., DFLL Humanities Bldg. 3, 1001 University Rd., Hsinchu 30010, Taiwan, courtney0419003.fl04@nctu.edu.tw)

Mandarin is a tone language with four phonemic tones (i.e., high-level Tone 1 [55]), rising Tone 2 [35], falling-rising Tone 3 [214], and falling Tone 4 [51] and with the maximum (C)(G)V(G)/(C) syllable structure. However, not all syllables can be combined with each of the tones (i.e., accidental gaps). For example, the syllable [ts^hu] is allowed to be combined with T1 ([ts^hu]55 "coarse"), T4 ([ts^hu]51 "vinegar"), but not T2 and T3. A calculation of all the 391 allowable syllables showed that there are 131, 185, 155, 110 accidental gaps in each of the four tones. A one-way chi-square test revealed that the accidental gaps in T2 were over-represented (stats here). A further investigation into these gaps in T2 according to different syllable types (CV: 52, CVN: 43, CGV: 38, and CGVN: 29) showed that these gaps were marginally under-represented in CGVN (stats here). We attributed these findings partially to the marked status of contour tones and to the typological preference for complex tonal targets to complex rimes due to their inherent longer durations (cf. Zhang 2001).

Session 4aSP

Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics: Detection and Tracking of Mobile Targets I

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

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Invited Papers

8:00

4aSP1. Detection and tracking in very shallow water. William Jobst, Lawrence Whited (Appl. Marine Phys., LLC, 1544 Marina Dr., Slidell, LA 70458, amp_jobst@bellsouth.net), and David Smith (Appl. Marine Phys., LLC, Miami, FL)

In our previous work, we demonstrated a low power, m-sequence based, bistatic acoustic system that detected and tracked a -20 dB target at a range of 172 m in a water depth of 2 m. To our knowledge, similar results have not been obtained with non-biological systems. Unfortunately, we were not able to repeat these results under similar, or possibly better, environmental conditions. This work addresses the reasons for failure to reproduce our original experimental results and suggests an approach that should result in very shallow water detection and tracking at much longer range.

8:20

4aSP2. Simultaneous estimation of array tilt and source range using broadband ship noise and a vertical array. Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu), Gihoon Byun (Scripps Inst. of Oceanogr., San Diego, CA), and Jea Soo kim (Korea Maritime and Ocean Univ., BUSAN, South Korea)

The array invariant, a robust approach to source-range estimation in shallow water, is based on the dispersion characteristics of broadband signals in ideal waveguides. It involves time-domain planewave beamforming using a vertical line array (VLA) to separate multiple coherent arrivals (eigenrays) in beam angle and travel time. Typically, a probe signal (i.e., a cooperating source) is required to estimate the Green's function, but the array invariant has been recently extended to a ship of opportunity radiating random signals using a ray-based blind deconvolution. Still, one major drawback is its sensitivity to the array tilt, shifting the beam angles and adversely affecting the array invariant parameter that determines the source range. In this paper, a simple optimization algorithm for simultaneous estimation of the array tilt and the source range is presented. The method is applied to a ship of opportunity (200–900 Hz) circling around a 56-m long VLA at a speed of 3 knots (1.5 m/s) at ranges of 1.8-3.6 km in approximately 100-m deep shallow water. It is found that the standard deviation of the relative range error significantly reduces to about 4%, from 14% with no compensation of the array tilt.

8:40

4aSP3. Tracking a surface ship via cascade of blind deconvolution and array invariant using a bottom-mounted horizontal array. Gihoon Byun, H. C. Song (Scripps Inst. of Oceanogr., La Jolla, San Diego, CA 92093-0238, gbyun@ucsd.edu), Jea Soo Kim (Korea Maritime and Ocean Univ., BUSAN, South Korea), and Ji Sung Park (Korea Inst. of Ocean Sci. and Technol., Busan, South Korea)

The array invariant developed for robust source-range estimation typically requires a priori knowledge about the source signal (i.e., cooperating source) to estimate the Green's function. However, the array invariant method has been recently extended to a surface ship radiating random signals (200-900 Hz) by extracting the Green's function via blind deconvolution using a vertical array [*J. Acoust. Soc. Am.* **143**. 1318–1325 (2018)]. In this paper, the blind deconvolution is applied to a 60-m long, bottom-mounted horizontal array to extract the Green's function of the same ship circling around the array in a square spiral pattern at ranges of 300–1500 m. The overall tracking performance shows good agreement with GPS measurements except when the ship is towards the broadside with respect to the horizontal array. Further, simultaneous localization of multiple ships is discussed.

Contributed Paper

9:00

4aSP4. Monitoring of moving surface ship using a four-element planar hydrophone array installed on the seabed. Sung-Hoon Byun and Sea-Moon Kim (KRISO, 32 1312beon-gil, Yuseong-daero, Yuseong-gu, Daejeon 34103, South Korea, byunsh@kriso.re.kr)

This talk presents the analysis result of underwater sound measurement data which were recently gathered in the shallow water near South Korea. For the underwater sound measurement, a four-element planar hydrophone array developed by KRISO was installed on the seabed and recorded a month-long acoustic data. The result shows that the measured sound level is usually higher in the morning time and this is ascribed to the surface boats which are presumably in fishing activity. We deal with the identification of the sound sources via various modulation analysis methods which can discriminate the sound induced by rotating propellers and also discuss the direction finding by beamforming of the planar hydrophone array data. [This work was financially supported by the research project PNS3120 funded by KIGAM.]

Invited Paper

9:15

4aSP5. Comparing passive localization methods for ocean vehicles from their underwater sound received on a coherent hydrophone array. Chenyang Zhu (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, zhu.che@husky.neu.edu), Delin Wang (Mech. Eng., Massachusetts Inst. of Technol., Boston, MA), Alessandra Tesei (Eng. Dept., Ctr. for Maritime Res. and Experimentation (CMRE), La Spezia, Italy), and Purnima Ratilal (Northeastern Univ., Boston, MA)

Simultaneous localization and tracking of multiple ocean vehicles over instantaneous continental-shelf scale regions using the passive ocean acoustic waveguide remote sensing (POAWRS) technique employs a large-aperture densely-sampled coherent hydrophone array system to monitor the underwater sounds radiated by ocean vehicles. Here, particle filtering is implemented for bearings-only localization and tracking of surface ships where the estimated bearing-time trajectories of ship-radiated underwater sound detections are used as inputs to provide two-dimensional horizontal position estimates. Coherent beamforming of the acoustic data received on the hydrophone array not only provides estimates of ship bearing, but also significantly enhances the signal-to-noise ratio. Results of passive acoustic surface ship localization and tracking from recordings in the Gulf of Maine and the Norwegian Sea are presented. The particle filtering approach for passive source localization are compared with three other approaches: moving array triangulation (MAT), array invariant (AI), and modified-polar-coordinates extended Kalman filter (MPC-EKF). The passive source localization accuracies, determined by comparison with the GPS-measured position of the surface ships, are dependent on numerous factors such as source-receiver geometry, range, and relative speed. By combining several of these approaches, which have respective pros and cons under different circumstances, the surface ships can be localized with improved accuracy.

Contributed Paper

9:35

4aSP6. Application of preprocessing methods to improve time delay estimation for synthetic aperture sonar motion estimation. Julia Gazagnaire (NSWC PCD, 2525 Pelican Bay Dr., Panama City Beach, FL 32408, jgazagnaire@hotmail.com) and Pierre-Philippe Beaujean (Ocean and Mech. Eng., Florida Atlantic Univ., Dania Beach, FL)

Synthetic aperture sonar (SAS) provides the best opportunity for side-looking sonar mounted on unmanned underwater vehicles to achieve high-resolution images at longer ranges. However, SAS processing requires maintaining a coherent phase history over the entire synthetic aperture, driving strict constraints on resolvable platform motion. This has driven the development of motion estimation and compensation techniques that use the received ping data, in addition to the onboard navigation solution, to resolve ping-to-ping platform motion. The most common approach is to use the redundant phase center technique. Here the ping interval is set, such that a portion of the array is overlapped. The accuracy of the motion estimation depends on the accuracy of the time delay estimation between the data received on the overlapping channels. Given the stochastic nature of the operational environment some level of decorrelation between these two signals is likely, even without residual platform motion. This decorrelation results in inaccurate time delay estimation and image quality degradation. In this research various preprocessing techniques have been applied to the sonar data to reduce the influence of stochastic noise with the goal of improving the accuracy of the time delay estimates.

Invited Papers

9:50

4aSP7. Mobile targets and sensor mobility in bat biosonar. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS Life Sci. District (Mail Code 0917), Virginia Tech, Blacksburg, VA 24061, rolf.mueller@vt.edu)

Many species of echolocating bats pursue highly mobile prey such as flying insects. Target mobility poses a challenge when a prey executes escape maneuvers. In these cases, bats and their prey find themselves locked in evolutionary arms races. However, prey mobility also provides the bats with opportunities for solving the problem of detecting prey buried in clutter. This applies to active as well as passive biosonar. In passive biosonar, bats frequently listen to sounds that are generated involuntarily by the motion of a prey. In active

biosonar, certain bat groups have evolved highly specialized biosonar systems to detect flying prey insects by virtue of unique Doppler signatures that are generated by the insect's wing beat. Since the Doppler shifts concerned are very small, this sensory feat requires a well-integrated suit of evolutionary specializations that include sonar signal design, cochlear filtering, representations all through the auditory system, and even adaptive behavioral control of the emitted frequencies. In addition to this complexity, bats with such a biosonar system also employ a peripheral mobility where the baffles for pulse emission and reception move rapidly during biosonar operation. It remains to be seen how these mobilities fit together to support the animal's sensory abilities.

10:10–10:25 Break

10:25

4aSP8. Underwater acoustic localization of sperm whales with a pair of hydrophones. Emmanuel Skarsoulis, George Piperakis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, N. Plastira 100, Heraklion GR-70013, Greece, eskars@iacm.forth.gr), Michael Kalogerakis (Technolog. Education Institute-Crete, Heraklion, Greece), Emmanuel Orfanakis, Panagiotis Papadakis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, Heraklion, Greece), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Alexandros Frantzis (Pelagos Cetacean Res. Inst., Vouliagmeni, Greece)

An iterative Bayesian approach using a pair of hydrophones was applied for the localization of vocalizing (click producing) sperm whales off the southern coast of Crete in the Eastern Mediterranean Sea in June 2015. The localization method relies on the time difference between direct and surface reflected arrivals at each hydrophone as well as between the direct arrivals at the two hydrophones, and on the knowledge of the hydrophones depths, whereas it accounts for refraction due to a depth-dependent sound-speed profile. The method provides range and depth estimates as well as estimates for the localization uncertainty reflecting measurement errors and environmental uncertainties. Further, by assuming a simple model for the array geometry bearing estimates were obtained. The inherent left-right ambiguity in bearing estimation can be resolved by changing the array orientation through maneuvering. The localization results were verified through encounters with the sperm whales when they ascended to the surface. [Work supported by EU-Greece through the Aristeia-II program and the NSRF 2014-2020/PERAN project.]

Contributed Paper

10:45

4aSP9. Tracking sperm whales *(Physeter macrocephalus)* detected with irregular time intervals. Tian Bai and Paul White (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton, Hampshire SO171BJ, United Kingdom, t.bai@soton.ac.uk)

Passive acoustic monitoring (PAM) is a method that has been proved to be a powerful tool for monitoring sperm whales. Sperm whales frequently emit loud, short duration, clicks for the purposes of echolocation. The clicks are emitted at intervals of typically once per second, but the exact interval between pulses varies irregularly. To achieve the monitoring goals, it is necessary to localise the vocalising animal. This is typically achieved by detecting the animal's echolocation clicks on a hydrophone array, estimating time delays between clicks and then employing a localisation algorithm. By adopting a tracking-based approach we are able to smooth the results, based on a model of whale motion, to remove noise and improve the estimates of location. The paper will consider the problem of tracking whales using the data from fixed bottom mounted sensors and using a tracking algorithm to follow the motion of such animals. The tracking will be implemented using a Kalman filter, but in this instance that filter needs to be designed to allow for data measured at irregular intervals. The method uses a version of the Kalman filter using a varying time-step. The time-step varying Kalman filter is applied to track data from simulated and measured datasets. The measured dataset is publicly available as part of the 2nd International Workshop on Detection and Localisation of Marine Mammals, Monaco, 2005 and comprises of a single animal moving within an array of 5 sensors.

Invited Paper

11:00

4aSP10. Comparing neural networks with conventional classifiers for fin whale vocalizations in beamformed spectrograms of coherent hydrophone array. Heriberto A. Garcia (Elec. and Comp Eng., Northeastern Univ., 151 Mystic St., Apt. 35, Arlington, MA 02474, garcia.he@husky.neu.edu), Seth Penna (Elec. and Comp Eng., Northeastern Univ., Boston, MA), Jess Topple (Sci. and Technol. Organisation - Ctr. for Maritime Res. and Experimentation (STO CMRE), La Spezia SP, Italy), and Purnima R. Makris (Elec. and Comp Eng., Northeastern Univ., Boston, MA)

A large variety of sound sources in the ocean, including biological, geophysical and man-made activities can be simultaneously monitored over instantaneous continental-shelf scale regions via the passive ocean acoustic waveguide remote sensing (POAWRS) technique by employing a large-aperture densely-sampled coherent hydrophone array. Millions of acoustic signals received on the POAWRS system per day can make it challenging to identify individual sound sources. An automated classification system is necessary to enable sound sources to be recognized. Here a large training data set of fin whale and other vocalizations are gathered after manual inspection and labelling. Next, multiple classifiers including neural networks, logistic regression, support vector machine (SVM) and decision tree are built and tested for identifying the fin whale and other vocalizations from the enormous amounts of acoustic signals detected per day. The neural network classifier will use beamformed spectrograms to classify acoustic signals, while logistic regression, SVM, and decision tree classifiers will use multiple features extracted from each detection to perform classification. The multiple features extracted from each detection include mean, minimum, and maximum frequencies, bandwidth, signal duration, frequency-time slope, and curvature. The performance of the classifiers are evaluated and compared using multiple values including accuracy, precision, recall, and F1-score.

Contributed Paper

11:20

4aSP11. Estimation of Shark's biomass through active acoustics. Roee Diamant (Marine Technologies, Univ. of Haifa, Rm. 280, Multi Purpose Bldg., 199 Aba Khoushy Ave., Haifa 3498838, Israel, roee.d@univ.haifa.ac.il), Eyal Bigal (Marine Biology, Univ. of Haifa, Haifa, Israel), Adi Pinhasi (Marine Technologies, Univ. of Haifa, Haifa, Israel), and Aviad Scheinin (Marine Biology, Univ. of Haifa, Haifa, Israel)

We study sharks biomass in open-sea using non-invasive active acoustics. The importance of continuous long-term monitoring of top-predator biomass is vital in understanding the healthiness of the ecosystem. Instead of the traditional fishery data, catch-and-release methods and visual inspections, which are problematic to supply reliable statistics, we rely on acoustic tools for the quantitative estimation of the number of sharks in a given area, their size, and the evaluation of their motion patterns and behavior. We take a blind classification approach and identify sharks' related reflections from sea boundary reflections based on a track-before-detect approach. Specifically, by emitting a series of wideband acoustic signals, we create a time-delay image whose rows correspond to the received reflection response. We rely on the observation that sea boundary reflections are characterized by a random clutter-like pattern, while shark's related reflections are continuous and steady. Thus, we detect a shark in a clutter by identifying in the time-delay image continuous but curved lines whose structure meet certain limitations, namely, the shark's maximal speed and its expected carangiformmotion pattern. In this paper, we will describe our method in details and show results from a sea experiment that included a verified detected shark.

THURSDAY MORNING, 8 NOVEMBER 2018 — OAK BAY 1/2 (VCC), 8:30 A.M. TO 11:45 A.M.

Session 4aUWa

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics: Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods I

Charles W. Holland, Cochair
Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804

Stan E. Dosso, Cochair
School of Earth & Ocean Sci, Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada

Chair's Introduction—8:30

Invited Papers

8:35

4aUWa1. Quantifying seabed and other environmental information contained in ocean ambient noise. Martin Siderius (ECE Dept., Portland State Univ., P.O. Box 751, Portland, OR 97207, siderius@pdx.edu) and John Gebbie (Adv. Mathematics Applications, Metron Inc., Portland, OR)

The ocean ambient noise field contains a surprising amount of information about the environment. Information about the seabed includes, layering structure, sound speed, density, and attenuation. There is also information about the ocean environment such as water column sound speed, volume attenuation, and the sea-state. Estimators for these quantities have been developed based mostly on vertical hydrophone arrays and beamforming. Although this has led to some simple and direct estimates for important quantities such as bottom loss, these estimates may be less than ideal in some cases. For example, when using inadequately sampled arrays or in certain unfavorable measurement geometries. The Fisher information can be used to quantify the basic information available in the noise measurements; and its inverse, the Cramér-Rao lower bound (CRLB), provides the lower limit on the variance of any unbiased estimator. The CRLB can be used to study the feasibility of various measurement configurations and parameter sensitivities. In this presentation, an overview will be given specifying the variety of environmental information contained in ocean noise. Parameter sensitivities and performance of beamforming estimators relative to the CRLB will also be discussed. [Work supported by the Office of Naval Research Ocean Acoustics Program.]

8:55

4aUWa2. Sequential filtering and linearization for inversion in the Seabed Characterization Experiment. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Using particle filtering we can track arrival times at spatially separated phones and identify them as distinct paths. This method is very well suited to data collected during the Seabed Characterization Experiment at 16 vertically separated phones of an MPL vertical line array. The source signals are mid-frequency linear frequency modulated pulses. Several paths can be identified in the received time-series including the bottom bounce and sediment reflection. These provide a plethora of information on the seabed properties. We combine the particle filter with a linearization approach that first allows us to resolve the geometry of the experiment. For that we build a Jacobian matrix with time derivatives with respect to the unknown parameters. We proceed with an exhaustive search for sediment parameters such as sound speed and thickness. Probability density functions of arrival times are propagated backwards through the inverse model, providing estimates of densities for the unknown parameters. [Work supported by ONR.]

9:15

4aUWa3. Insights into sediment acoustics from analytic solutions of forward problems. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Closed-form analytic solutions of wave equations are available for a rather limited subset of geoacoustic models of practical interest. Apart from elucidating the challenging underlying physics of wave propagation, the analytic solutions prove to be useful in sediment acoustics when these predict distinct, identifiable features of measured acoustic fields. Theoretical predictions can be used then to identify or constrain the type of sediment stratification, lead to quick, low-parameter inversions, guide application of statistical inversion methods, and help with interpretation of their results. In this paper, application of analytic solutions of forward problems to acoustic remote sensing of marine sediments will be illustrated by results obtained in a few experiments, which involve observations of slow interface waves near the seafloor and/or resonance peaks in either bottom-reflected energy or in the power spectrum of ambient noise. Coupling between shear and compressional waves in the stratified ocean bottom plays a key role in both examples. [Work supported, in part, by ONR.]

9:35

4aUWa4. Trans-dimensional geoacoustic inversion on the New England Mud Patch using modal dispersion data. Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Dimitrios Eleftherakis (ENSTA Bretagne, Brest, France), N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

This paper presents trans-dimensional geoacoustic inversion of two low-frequency broadband signals recorded during the Seabed Characterization Experiment (SBCEX). The considered sources are a chirp emitted by a towed J15 source and an underwater impulse created by a combustive sound source (CSS). Corresponding received signals are recorded on single hydrophones, and the two source/receiver configurations are reciprocal, on the same track. Input data for inversion are modal time-frequency dispersion curves, estimated using warping. Inversion is then performed within a Bayesian framework. A trans-dimensional inverse algorithm is used to estimate the model parametrization (i.e., number of seabed layers), and the mode covariance matrices are estimated as a first-order autoregressive error process. Inversion results on the two reciprocal tracks are consistent with the modal information available in each dataset. Results are also consistent with what is known about the area.

9:55

4aUWa5. Application of geoacoustic inference for ecosystem monitoring of a seagrass meadow. 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), Gabriel R. Venegas (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Abdullah Rahman (Sch. of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, South Padre Island, TX), and Justin T. Dubin (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Seagrasses provide a multitude of ecosystem services: they alter water flow, cycle nutrients, stabilize sediments, support the food web structure, and provide a critical habitat for many animals. However, due to threats to seagrass meadows and their associated ecosystems, these habitats are declining globally. Since the biological processes and physical characteristics associated with seagrass are known to affect acoustic propagation due to bubble production, which results in dispersion, absorption and scattering of sound, acoustical methods are proposed to assess the health of seagrass meadows. For this purpose, an experiment was conducted in the Lower Laguna Madre where the seabed was covered by a dense growth of Thalassia testudinum. During the experiment, a combustive sound source was used to produce broadband signals at ranges of 20 m to 1000 m from the receiver location. Three sensors were positioned at the receiver location: two hydrophones located within and above the seagrass canopy, and a single-axis geophone. The data were analyzed for the purposes of inferring environmental parameters in the seagrass meadow and to investigate the feasibility of using acoustical methods to monitor ecosystem health. Initial results indicate that water column void fraction can be inferred. [Work sponsored by ARL:UT IR&D and ONR.]

10:15–10:30 Break

Contributed Papers

10:30

4aUWa6. Geoacoustic parameter variations and inversion estimates with a silt-suspension theory of mud. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., Mathematical Sci., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., East Sandwich, MA), Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jason D. Chaytor (Coastal and Marine Sci. Ctr., U.S. Geological Survey, Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Marine mud sediments can be modeled by a recent silt-suspension theory [Pierce, *et al.*, *JASA*, **142**, 2591 (2017) (A)], in which silt particles are embedded in a suspension of flocculated clay particles. This presentation investigates the influence on attenuation predictions from uncertainties in parameters such as the effective grain density and the distribution of grain sizes. For example, the matrix of clay flocs may effectively lower the density of silt grains and cause decreased attenuation. Effects from such parameters are determined on the regime where attenuation is nearly linear with frequency. The value of porosity is critical for specifying sound speed and attenuation in the mud layer. Porosity measurements from sediment cores can have significant uncertainty, particularly in the upper region, from the core extraction process. Inversions for porosity and other physical parameters of the silt-suspension theory are performed along a WHOI AUV track from the 2017 Seabed Characterization Experiment. Characteristics of cores guide parameter modeling and ranges for the inversions. This approach produces estimates of sound speed, attenuation, and density that are related by physical constraints. In addition, the approach allows for validity assessment of current geoacoustic mud models using measured data. [Work supported by ONR.]

10:45

4aUWa7. Maximum likelihood geoacoustic inversion from surface generated noise. John Gebbie (Adv. Mathematics Applications, Metron, Inc, 2020 SW 4th Ave., Ste. 2020, Portland, OR 97201, gebbie@metsci.com) and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

Noise generated at the surface from wind and breaking waves is incident upon the seabed at a wide range of angles and frequencies. The reflectivity of the seabed thus plays an important role in determining the vertical directionality of the noise field. It is therefore not surprising that from an information theory perspective, the noise field encodes a significant amount of information about the geoacoustic properties of the seabed that govern reflectivity, such as layering structure, sound speed, density, and attenuation. Using a physics-based ambient noise model, the Cramér-Rao lower bound (CRLB) can be computed, which specifies the lowest possible variance that an unbiased estimator can have. Existing estimators have primarily involved vertically beamforming the noise field to recover the vertical noise directionality and bottom loss, which is then processed to estimate geoacoustic seabed properties. The method described here takes a different approach by estimating the properties directly from the measured data (i.e., without beamforming). This presentation will show that maximizing the proposed likelihood function can perform better than beamforming-based techniques, and approaches the CRLB. Further, it remains unbiased in low signal-to-noise ratio conditions, is tolerant to array tilt, and can operate beyond the nominal array design frequency.

11:00

4aUWa8. Classification of acoustic and seismic signals based on the statistics of their wavelet sub-band coefficients. Costas Smaragdakis (Dept. of Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Heraklion, Greece), John Mastrokalos (Inst. of Appl. and Computational Mathematics, FORTH, Heraklion, Greece), and Michael Taroudakis (Dept. of Mathematics and Appl. Mathematics & IACM, Univ. of Crete and FORTH, Voutes University Campus, Heraklion 70013, Greece, taroud@uoc.gr)

Statistical characterization of acoustic signals is a pre-processing technique aiming at the definition of signal observables, which can be used as input data in the formulation of inverse problems of acoustical oceanography aiming at the estimation of critical parameters of the marine environment. Statistical signal characterization is also a means to classify the signals and compare their properties without reference to any physical model that determines the signal observables. Thus, it can in principle be used for the comparison of the signals of any type, including seismic recordings, thus opening the area to many applications of geophysical monitoring, using signals of any type. The paper summarizes the first attempts to study the efficiency of a signal characterization method based on wavelet transform of the signal at various levels, followed by the statistical description of their sub-band coefficients. It is shown that A-stable symmetric distributions are capable of defining the statistics of these coefficients for signals used in applications of ocean acoustic tomography and geoacoustic inversions. It is further investigated if these distributions are capable of defining the statistics of seismic signals and it is shown by preliminary results, that they can indeed be considered as possible candidates in this respect.

11:15

4aUWa9. Estimation of subbottom geoacoustic properties from offshore airgun surveys in Atlantic Canada. Graham A. Warner (JASCO Appl. Sci., 2305-4464 Markham St, Victoria, BC V8Z 7X8, Canada, graham.warner@jasco.com) and Bruce Martin (JASCO Appl. Sci., Dartmouth, NS, Canada)

This paper estimates seabed geoacoustic profiles at 14 sites offshore of the eastern Canadian coast using sound levels from a single airgun. A 210 in^3 airgun was operated along two track lines at each site and a calibrated seabed-mounted Autonomous Multichannel Acoustic Recorder equipped with a single omnidirectional hydrophone recorded the airgun sounds. High-resolution bathymetry was acquired along the tracks using a ship-mounted multibeam system and conductivity-temperature-depth profiles were measured at each site. Seabed geoacoustic profiles were estimated in a non-linear trans-dimensional Bayesian inversion using energy spectral density levels at six frequencies between 10 and 320 Hz as a function of range. The inversion estimated the unknown number of subbottom layers, geoacoustic properties (including compressional and shear wave speeds and attenuations, density, and layer thicknesses), airgun source levels, a range-correction factor, and error statistics. Data were fit with error standard deviations of 0.6–5.6 dB. Convergence of the posterior probability density function was not reached within the time limits of the study so parameter uncertainty was estimated using a linearized formulation of the inverse problem centered around the model that minimized the Bayesian Information Criterion.

11:30

4aUWa10. Classifying seabed parameters from normal incidence reflections. Megan Frantz, Martin Siderius (ECE, Portland State Univ., P.O. Box 751, Portland, OR 97207, mefrantz@pdx.edu), and Scott Schecklman (Adv. Mathematics Applications, Metron, Inc., Portland, OR)

A technique is being investigated to estimate and classify seabed parameters, including interface roughness, based on normal incident reflections. These parameters can be characterized by single beam echo sounder data or normal beams from a side scan sonar. Previous sediment classification methods used either empirical relationships based on grain size (estimated based on the strength of the echo return) or by matching data with a parameterized model of the echo envelope. [Snellen, Siemes, Simons. J Acoust Soc Am. 2011]. Here, seabed classification (including interface roughness) is estimated using a combined approach. Both the strength of the reflection and a model of the echo envelope are used to determine seabed properties. The modelled envelope is obtained from the parameterization of the scattering cross section derived from the high-frequency Kirchhoff approximation near vertical incidence [Jackson, Richardson. High-Frequency Seafloor Acoustics]. To determine the validity of the modelling approximations, a merit function has been developed to quantify the similarity between the data returned and the model results. In this presentation, the model-based approach to classifying seabed properties will be described along with results showing the exact and approximate scattering results. [Work supported by the Office of Naval Research.]

THURSDAY MORNING, 8 NOVEMBER 2018 SAANICH 1/2 (VCC), 9:00 A.M. TO 11:40 A.M.

Session 4aUWb

Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Acoustic Vector Field Studies I

Kevin B. Smith, Cochair
Department of Physics, Naval Postgraduate School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943

Robert Barton, Cochair
NUWC, 1176 Howell St, Newport, RI 02841

Chair's Introduction—9:00

Invited Papers

9:05

4aUWb1. Some comments on acoustic sensing at a single point in space. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Camille Pagniello, and Timothy Rowell (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA)

The directionality provided by an acoustic vector sensor can be viewed in the context of a Taylor series expansion of the acoustic pressure field. From this perspective, acoustic pressure and acoustic particle velocity are but two of an infinite number of acoustic field variables. The purpose of this presentation is to examine some of the properties of these higher order field variables, including their energetics and the behavior of their spatial cross spectra in homogeneous, isotropic noise. Analytical expressions for those spatial cross spectra involving the normal strain rate parallel to the direction of separation decrease rapidly with increasing separation, and then continue to oscillate with a sinc function behavior. In contrast, those involving the normal strain rate perpendicular to the direction of separation decrease more slowly so that significant positive coherence exists at half-wavelength spacing, but then decay to, and remain near, zero at larger spacings. Accurate estimates of the range to a near-field point source can be obtained from the in-phase and quadrature components of normal strain rate with respect to pressure. Do animals in the ocean exploit the properties of these higher-order field variables?

9:25

4aUWb2. Exploiting ambient noise for coherent processing of mobile vector sensor arrays. Karim G. Sabra, Brendan Nichols, James S. Martin (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu), and Christopher M. Verlinden (US Coast Guard Acad., La Jolla, CA)

A network of mobile sensors, such as vector sensors mounted to drifting floats, can be used as an array for locating acoustic sources in an ocean environment. Accurate localization using coherent processing on such an array dictates the locations of sensor elements must be well-known. Achieving this for a mobile, submerged array composed of individual drifting sensors is usually challenging. However the coherent processing of the ambient acoustic noise between sensor pairs can provide the separation distance between them and thus an opportunity to correct sensor location errors. Here a stochastic search algorithm is presented for identifying hidden coherent noise arrivals when the separation distance is changing faster than the required averaging time to extract such arrivals from a fixed sensor pair. The accuracy of this method matches that of GPS-derived array element positioning and its performance is shown to be improved when using directional vector sensors instead of omnidirectional hydrophones. The proposed approach is demonstrated

4a THU. AM

experimentally using ambient noise recorded by drifting vector sensors deployed in the Long Island Sound and is used to enable tracking of surface ships by coherent processing of the drifting sensors.

9:45

4aUWb3. Characteristics of very-low-frequency pulse acoustic fields measured by vector sensor and ocean bottom seismometer in shallow water. Renhe Zhang, Shihong Zhou, Yubo Qi, Yuquan Liang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Bei-si-huan-Xi Rd. No.21,Haidian District, Beijing 100190, China, shih_zhou@mail.ioa.ac.cn), and Yuanjie Sui (Qingdao Branch, Inst. of Acoust., Chinese Acad. of Sci., Qingdao, China)

In the marine environment, the effects of the seafloor and subbottom elastic media on the propagation of very low frequency (VLF) sound wave should be considered. In order to understand the mechanism of seismoacoustic field propagation, a very-low-frequency sound propagation experiment was conducted in shallow water using seafloor-located acoustic vector sensor and ocean bottom seismometers (OBS) in December 2017. The water depth is about 100 m. Pulse sources detanoted at depths of 7 m and 50 m were employed to emit the VLF signals. Comparing the signals received by OBS with that of the acoustic vector sensor, a narrow-band wave package group at VLF had been observed, which was excited by the interaction of acoustic waves in water with the layered shear seabed. In this paper, the pulse signals are used to analyze the characteristics of sound propagation and spatial correlation, amplitude and phase structure of separated normal modes based on warping transform, horizontal and vertical acoustic energy flow at different ranges. The time arrival structure and time expansion of the elastic-bottom-induced wave package group and its relationship with source distance and depth are also analyzed.

10:05

4aUWb4. A primer on vector hydrophones. Bruce A. Armstrong (GeoSpectrum Technologies Inc., 10 Akerley Blvd., Unit 19, Dartmouth, NS B3B 1J4, Canada, bruce.armstrong@geospectrum.ca)

Vector hydrophones are used by the military to track submarines, and, more recently, by scientists attempting to measure the influence of particle velocity on the behaviour of fish. Despite being in use for decades, vector hydrophones remain poorly understood by both communities. The problem is particularly acute for those aiming to measure particle velocity because vector hydrophones do not measure particle velocity directly; rather they measure pressure difference, the force that creates particle velocity. Topics covered will be the two types of vector hydrophone and why only one works well at low frequencies, the importance of knowing the phase of the directional channels relative to each other and to the pressure hydrophone, how bearings are determined, calibration of vector hydrophones and near-field effects, the magnitude of particle velocity for typical sound pressure levels, why bigger is better for mechanical noise, why neutral buoyancy is unnecessary and undesirable, and best practice for mounting.

10:25–10:40 Break

Contributed Papers

10:40

4aUWb5. Frequency warping transform of the vertical energy flux and its use for passive source ranging. Yubo Qi, Mengxiao Yu, Shihong Zhou, Renhe Zhang, Shuyuan Du (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Bei-si-huan-Xi Rd., No.21, Haidian District, Beijing 100190, China, qyb@mail.ioa.ac.cn), and Mei Sun (School of Phys. and Electron. Eng., Taishan Univ., Taian, China)

The frequency warping transform of the vertical energy flux and a passive impulsive source ranging method with a single acoustic vector sensor are presented in this paper. The cross-correlation component of two different modes in the vertical energy flux is transformed into separable impulsive sequence. With a guide source, the source range is extracted from the relative delay time of the impulsive sequence. Comparing with warping the pressure autocorrelation function, there is no need to delete the modal autocorrelation component for warping the vertical energy flux, especially for its real part. Besides, the source ranging result based on the time delay of the warped vertical energy flux is much better for closer range. The frequency warping transform of the vertical energy flux and the passive source ranging method are verified by experimental data.

10:55

4aUWb6. A bottom-diffracted surface-reflected arrival in the North Pacific observed from 15 to 3200 km. Ralph A. Stephen (RASCON Assoc. LLC, PO Box 567, West Falmouth, MA 02574, rasconassoc@aol.com), S. T. Bolmer (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew 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. They were first identified in the LOAPEX Experiment in the North Pacific in 2004 where a BDSR from the side of “Seamount B” was observed from 500 to 3200 km range for an M-sequence centered at 75 Hz. In a follow-up research cruise in 2013, a BDSR was also observed from the side of “Seamount B” for ranges from 15 to 35 km and the same azimuth. Are the seafloor diffractors observed in 2004 and 2013 the same? Various measures of the diffractor location and their resolution are reviewed. Both diffractors fall within a 1km radius region. Elsewhere in the 2013 survey pairs of BDSRs were observed within a kilometer of each other. So it is possible that the two BDSR arrivals arise from two distinct and unresolvable diffractors. It is quite likely however that the two diffractor points are the same. The location of the 2004 diffractor is sufficient to explain the arrival times of the BDSR from the 2013 diffractor within the scatter of the data. [Work supported by ONR.]

11:10

4aUWb7. Underwater vector sensor communication in KOREX-17. Kang-Hoon Choi, Sunhyo Kim, Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, choikh0210@hanyang.ac.kr), Su-Uk Son (The 6th R&D Inst., Agency for Defense Development, Chanwon-si, Gyeongnam, South Korea), Peter H. Dahl, David R. Dall’Osto (Appl. Phys. Lab., Univ. of Washington, USA, Seattle, WA), and Ho Seuk Bae (The 6th R&D Inst., Agency for Defense Development, Changwon, South Korea)

The underwater acoustic communication channel is subject to multiple interactions with the ocean boundaries and refraction due to sound speed structure of water column, which produces significant time spread. This phenomenon is referred to as inter-symbol interference (ISI), which results in degradation of error performance. Recently, a time reversal technique has been

used for reducing the ISI. However, this requires a large-size array with spatially separated receivers to obtain higher spatial diversity gain, and it becomes a limitation to its application in space-constrained environment. An acoustic vector sensor (combined pressure and particle velocity) can potentially yield a better communication performance relative to a system using only hydrophones. In this talk, communication data collected using a vector sensor known as IVAR (Intensity Vector Autonomous Receiver) during Korea Reverberation Experiment (KOREX-17) conducted in shallow water located at 34° 43′ N, 128° 39′ E on May 23–31, 2017. The characteristics of the channel as probed by a pressure-only versus combined pressure plus particle velocity system are discussed along with some performance results of a vector sensor communication system. [Work supported by the ADD(UD170022DD) and the National Research Foundation of Korea(NRF-2016R1D1A1B03930983).]

11:25

4aUWb8. Implementation of Nuttall's vector-sensing target motion analysis algorithm. Benjamin Cray (NUWC, 1176 Howell St., Newport, RI 02841-1708, benjamin.cray@navy.mil)

Direction cosine measurements, obtained from an array of vector sensors, are processed and used to estimate the location and motion of a moving source. Nuttall derived this unpublished TMA algorithm in 2006, which recently was reviewed, implemented, and evaluated using newly collected experimental data. Both stationary and moving sources are examined. It is assumed that the array element positions are known; however, at any given time instant, the measured direction cosines are presumed noisy. Originally, Nuttall's formulation only utilized the three orthogonal components of acoustic particle velocity at each array location, however adding acoustic pressure is straight-forward and resolves bearing ambiguity. For a non-moving source, the TMA processor (linear in source position variables) generates a 3-by-3 set of simultaneous equations, with solution vector for the source position (x_s, y_s, z_s). Range from any array element to the source can then be estimated. For a moving source, a 6-by-6 set of simultaneous equations is obtained; the solution vector is now the initial source position (x_s, y_s, z_s) and the source velocity components (a, b, c). It is presumed that the source velocity is constant over the observation interval.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 THEATER (VCC), 1:15 P.M. TO 4:50 P.M.

Session 4pAA

Architectural Acoustics and Noise: Validation of Modeling and Analysis: Predictions and Outcomes

Logan D. Pippitt, Cochair
none, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045

Ben Bridgewater, Cochair
Architecture, University of Kansas, 1536 Ogden Street, Denver, CO 80218

Invited Papers

1:15

4pAA1. Practical use and extension of methods for improved source realism in auralizations. Marcus R. Mayell, Nicolaus Dulworth, and Gregory A. Miller (Threshold Acoust., LLC, 141 West Jackson Blvd., Chicago, IL 60604, mmayell@thresholdacoustics.com)

The effectiveness of an auralization as a learning and decision-making tool is greatest when listeners are able to actively engage in a plausibly realistic experience. Accuracy of the spatially varying timbral behavior of various sound sources within acoustic models has been found to greatly improve the perceived realism in auralizations. Multi-channel/phantom source modeling techniques offer the ability to better emulate sources through improved timbral representation, more authentic 3D imaging, and dynamic directivity—characteristics limited or lost by traditional point source representation methods. This technique realizes and expands upon methods using multichannel recording techniques that have been previously proposed. It offers the ability to expedite the source capture and implementation process, and to more easily represent larger/multiple sound sources while improving the feasibility and opportunity for high quality source content.

1:35

4pAA2. Speech Privacy Class calculations and post-construction measurements of a corporate workplace tenant improvement project. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com)

LEED interior design and construction projects IEQ acoustical credits are particularly difficult to achieve due to broad scope requirements and restrictive credit language. Specifically, the high composite sound transmission requirement between private offices and hallways. DLR Group acoustical staff proposed an alternative compliance path and evaluation focused on lower STC rated partitions with the addition of sound masking systems to increase the background noise levels, with the goal of achieving an equivalent level of speech privacy between offices and hallways. Speech Privacy Class as detailed in ASTM E2638-10 was used as the evaluation metric. Difficulties in analysis and design stage assessment included unknown level difference of constructed partitions and entry doors, correlating subjective Client expectations to objective SPC ratings, and Client sign-off on sound masking approach. Post-construction measurements were conducted before and after sound masking system installation. Significant improvements in objective SPC ratings were measured, although targeted design goal SPC ratings were not achieved in all cases. Measured noise level differences of the glass storefront office partitions and measured sound masking noise spectra were compared to design stage values for further process refinement. Completed space measurement results and design stage comparisons will be reviewed and discussed.

1:55

4pAA3. A tale of three models, two loudspeakers, and one install. Liz Lamour Croteau (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776, elamour@cavtocci.com)

It's commonplace to use an acoustic modeling software to confirm room conditions and loudspeaker coverage when designing a sound system. It is also becoming more commonplace for different loudspeaker manufacturers to publish—and require—the use of their own "homemade" proprietary software for modeling. In this survey, three separate software applications were utilized to model two loudspeaker options for a lecture hall at a local university with one loudspeaker solution ultimately being selected and installed. The university required graphic evidence of adequate coverage from both solutions and an on-site loudspeaker "shoot out" to determine final loudspeaker selection. Data collected along the journey includes room coverage maps for each loudspeaker with graphic elements adjusted for close-to-equal comparison, rough pink noise octave-band measurements during the demo, and final installed analysis and comparisons.

2:15

4pAA4. Learning about hall acoustics from multi-directional audio recordings. Benjamin Markham, Jonah Sacks, and Kelsey Hochgraf (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Multi-channel audio recording, for playback in ambisonic format, is common when documenting a newly completed auditorium or an existing auditorium to be renovated. These recordings provide opportunities for comparative listening of different auditoria, and also between an acoustical model and the built auditorium. To allow meaningful comparison, recordings have been made using the same musicians playing the same musical excerpts on the same instruments. Anechoic recordings made similarly are used for convolution with synthetic (modelled) impulse responses. Comparative listening experience has yielded a number of lessons for acoustical design of auditoria and for acoustical modeling in design, including (a) the importance of instrument directivity to the sound of music in an auditorium, and perceptual responses to variations thereof, (b) the importance and challenge of loudness calibration during comparative listening, (c) inter-dependency of room size, perceived intensity and distribution of reverberation, and source directivity, (d) limitations in typical geometric acoustical modeling techniques, and (e) usefulness of directive loudspeakers when acquiring measured impulse responses for the purpose of convolving with anechoic audio. The process and results of these efforts will be presented and discussed. Selected listening segments will be available following the presentation.

Contributed Papers

2:35

4pAA5. Method of predicting Composite Sound Transmission Class rating of composite wall assemblies that do not extend to deck. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA , spettyjohn@acousticsandvibration.com)

Predicting the sound transmitted through a partition can be very difficult because of the range of unknowns. Typically, Sound Transmission Class, STC, data provided by vendors is available to select the appropriate wall or floor/ceiling assembly. However, when partitions are made of multiple building elements, the difficulty increases. Calculating the area of each element and having an STC rating for each element allows the Composite STC rating to be calculated from the total area divided by the Total area times the transmission loss coefficient. The larger the STC rating, the smaller the Transmission Loss Coefficient. Thus, even small areas with very low STC ratings diminish the Composite STC rating. For a wall partition that stops 15 cm above the ceiling, the area and STC rating of the common area must be included in the calculation. This means that the area of the ceiling, the penetrations of the ceiling, and the area from the top of the partition must be computed. A method has been developed to account for all the paths between two spaces that provides a good estimate of the Composite STC/NIC rating of any partition. Examples and details will be provided.

2:50

4pAA6. Regression and correlation analysis of measured and simulated reverberation time in two Taiwanese Basilica churches. Chia-Fen Lee and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., 3F., No. 87, Ningbo W. St., Zhongzheng Dist., Taipei City 10075, Taiwan, rebecca830111@gmail.com)

There are few historical Basilica churches in Taiwan. None of them have been studied thoroughly on their acoustic characteristics with on-site measurements and model predictions. Wanchin Basilica of the Immaculate Conception and Tainan Theological College and Seminary Chapel were built between 1860 to 1960 AD. On-site measurement with ISO-3382 were performed to obtain reverberation time and background noise level. To obtain and document additional acoustic parameters and mappings of C_{80}, D_{50}, and STI, Odeon Acoustics was used. Since both churches are the historical buildings, it is difficult to identify the absorption coefficient of materials used in these churches for model simulation purpose. Thus, after selecting the approximate materials for the simulated model, validating the measured and simulated reverberation time with Linear Regression model is necessary. With 392 data points; standard error, least squares, and RMSD were calculated to derive the regression model with correlation (R) and determination coefficient (R^2). R^2 of regression models for both churches are 0.73 and 0.84. R values for both churches are 0.86 and 0.92. Both R and R^2 indicated that the high correlation between predicted model and measured data. Thus, the confidence of predicated C_{80}, D_{50} and STI can be raised and reliable.

3:05–3:20 Break

3:20

4pAA7. Compatibility study between building information modeling and acoustic simulation software. Kaveh Erfani, Sara Mahabadipour, Joonhee Lee, and Mazdak Nik-Bakht (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada, joonhee.lee@concordia.ca)

Geometric and non-geometric data from Building Information Models (BIM) need to be manually reconstructed in acoustic simulation software to examine building acoustic performance. This process is time-consuming, error-prone, and the accuracy of acoustic analysis results will depend on the expertise of the user. The process is also unidirectional (i.e., from BIM to

the acoustic software); meaning that simulation outputs can't be easily retrieved back to the BIM. Thus, over the last decade, several attempts have been made to integrate acoustic analysis into BIM-related software during the conceptual design phase. This paper aims to facilitate the interoperability between BIM and acoustic simulation engine by improving the information exchange between them. A case study with an educational space at Concordia University in Montreal is carried out to validate the developed system and examine the practicality and efficiency of this process in action. The paper will investigate: 1) extraction of geometry data from the BIM software; 2) enriching the BIM with data regarding acoustic absorption coefficients, via an external database; 3) calculation of acoustic qualities including reverberation time, and 4) visualization of the simulation results in the BIM software.

3:35

4pAA8. Sound insulation of a single finite panel—Comparison, validation, and parametric studies. Yihe Huang, Fangliang Chen, and Tejav DeGanyar (Virtual Construction Lab, Schuco-USA, 260 W 39th St., Ste. 1801, New York, NY 10018, yhuang@schuco-usa.com)

Sound transmission characteristics of a single finite panel is of significant value for both theoretical development and engineering application in building acoustics. Extensive studies and numerous models have been developed to study the sound transmission loss behavior of a single panel over the past decades. Though most of them have been validated separately with specific test results in their developing phases, evidences showed that considerable discrepancy exit from one model to another. Numerous comparisons among different prediction models have been presented in the literature. However, quantitative comparisons among them is still lacking. To guide the design of real practice, an impartial evidence of the applicability and accuracy of different models is highly desired. For this purpose, a quantitative comparison among more than ten widely adopted existing models are presented in this study. Test results collected from different labs for different materials, that found in the literature and conducted by the authors as well, are provided first to validate different models for different scenarios. Comprehensive parametric studies are then conducted to investigate the effects of different design parameters, such as boundary conditions, damping factor, panel dimension, and aspect ratio, on the sound transmission loss of a single panel.

3:50

4pAA9. Basic study on three-dimensional sound field auralization using a scale model. Kazunori Suzuki, Yoshinari Yamada, Shinichiro Koyanagi, and Takayuki Hidaka (Res. and Development Inst., Takenaka corp., 1-5-1, Ohtsuka, Inzai-shi, Chiba 270-1395, Japan, suzuki.kazunori@takenaka.co.jp)

In order to achieve ambisonic three-dimensional sound field auralization using a scale model, the ambisonic recording procedure was reformulated based on a partly modified microphone arrangement. Specifically, a technique of measuring room acoustic response by changing a small microphone to various places in a scale model was introduced. In converting room acoustic responses to ambisonic signals, taking signal differences reduces the dynamic range of the low frequency range and deteriorates the sound quality of auralization. This is avoided by determining the microphone arrangement interval and measurement frequency range with the characteristics of the model sound source and the microphone SN ratio taken into consideration. The room acoustic responses measured in the divided frequency range was synthesized to obtain all the frequency range. To confirm the effectiveness of this method, ambisonic signals up to the secondary ones were prepared, and a fundamental psychological experiment on localization in the horizontal plane was conducted, and localization reproducibility equivalent to a full-scale model was confirmed. The presented sound contains no auditorily harmful noise and is expected to be applicable to subjective evaluation of the sound field of a hall.

4:05

4pAA10. Measurement and simulation of the sound field in a concrete grain silo. Daniela T. Helboe (Acoust., COWI AS, Karvesvingen 2, Oslo 0579, Norway, datl@cowi.com)

This paper presents impulse response measurements of a concrete grain silo and results from simulations of the measured sound field. The purpose of the measurement and simulations has been to assess the validity of the evaluation tools used for the acoustic design of an art museum, a project comprising the renovation of an existing reinforced concrete grain silo building from 1935 into a space for exhibition of art. The architects' concept involves cutting the lower half of the silos to create a central space (approximate volume 6500 m^3, height 28 m) and exhibition spaces (approximate volume 2600 m^3, average height 12 m), while keeping the upper half of the cylinders with an open end to these spaces. This paper presents how the measurements and simulations of a single concrete grain silo (approximate volume 450 m^3, height 30 m) were used to adjust relevant settings in the acoustic simulations. Uncertainties related to the simulation tools are presented and acoustical challenges imposed by the geometry of the proposed architectural concept are discussed.

4:20

4pAA11. Public lobby reverberation control using modelling software design. Ellen R. Buchan and Philippe Moquin (TSB, AB Infrastructure, 6950-113 St. NW, Edmonton, AB T6H5V7, Canada, ellen.buchan@gov.ab.ca)

A public lobby with high ceilings and hard surfaces is often found to be an excessively reverberant space. Sound-absorbing finishes to ensure a good acoustic environment are required. This case study examines the acoustical analysis of the public lobby and adjacent corridors at the Edmonton Law Courts. Improving the acoustics of this workplace/public space is important to create comfortable working conditions, and minimize noise disturbances in the nearby courtrooms. Design parameters included reverberation time and sound pressure level (SPL). ODEON Auditorium software was used to determine the optimum sound-absorbent finishes required. An array of acoustical ceiling panels and wall panels were installed in the Lobby from this simulation. The results of acoustical measurements conducted after the acoustical renovation and the ODEON Modelling predictions are discussed and compared.

4:35

4pAA12. On-stage energy enhancement in a vineyard concert hall verified by scale model measurements. Anh-Nguyen Phan, Weihwa Chiang, and Yi-Run Chen (College of Design, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd., Section 4, Taipei 106, Taiwan, whch@mail.ntust.edu.tw)

Reflective surface area near the stage of a vineyard concert hall is largely reduced when surrounded by seating terraces as opposed to by perimeter walls in a rectangular hall. Lacking of upper wall surfaces would dramatically reduce reflective energy directed back to the performer himself and to the others. The missing energy could, however, be enhanced by reflective energy from overhead, detached reflectors and other terrace walls in front of the stage. Computer simulations and scale model measurements are conducted considering tilting of side stage walls, form of the overhead reflector and height of the overhead reflector. The enhancement in both early support and reflective sound strength on stage is roughly 2.5 dB. Making the reflector surface highly diffusive causes slight drop in early support but increase in hall-averaged sound strength.

Session 4pAB

Animal Bioacoustics and Signal Processing in Acoustics: Combining Passive and Active Acoustics for Ecological Investigations

Simone Baumann-Pickering, Cochair
Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Ana ÕiroviÃ, Cochair
Scripps Institution of Oceanography, 9500 Gilman Drive MC 0205, La Jolla, CA 92093-0205

Chair's Introduction—1:30

Invited Paper

1:35

4pAB1. The ocean has music for those who ping AND listen. Kelly J. Benoit-Bird (Monterey Bay Aquarium Res. Inst., 104 COEAS Admin Bldg., Corvallis, Oregon 97331, kbb@mbari.org), Brandon L. Southall (Southall Environ. Assoc., Inc., Aptos, CA), and Mark Moline (Univ. of Delaware, Lewes, DE)

Sound is a key sense for animals in the ocean, including humans. The integration of passive and active acoustics in behavioral and ecosystem studies has revealed much about how ocean systems work. Here, I will discuss how the integration of echosounders with various passive acoustic approaches has contributed to our understanding to the biology and ecology of marine mammals. For example, explaining the habitat selection of beaked whales, the foraging ecology, behavior, and communication of spinner dolphins, and the ability of dolphin sonar to discriminate fish species. Careful integration of passive and active acoustics has also contributed to sonar signal design and processing approaches. A recent study of Risso's dolphins using passive acoustic recording tags on the dolphins themselves and echosounders deployed from a ship and an underwater vehicle provided new insights into how these animals make foraging decisions, what information is available to them and when, and how changes in their prey over space and time affect their foraging behavior. The combination of passive and active acoustics is providing insight into the ecology and dynamics of predator and prey at the scale of the individual and the population.

Contributed Papers

1:55

4pAB2. Sonar signal analysis: Biological consequences of out-of-band acoustic signals from active sonar systems. Paul A. Lepper (Wolfson School, Loughborough Univ., Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk) and Denise Risch (Scottish Assoc. of Marine Sci., Oban, Argyll, United Kingdom)

Within the aquatic environment acoustic monitoring systems are widely used to monitor and assesses anthropogenic noise sources and potential impacts on many aquatic species. Passive acoustic studies are, however, often limited to acoustic observation of changes in the species vocalization. Changes in vocalizations such as vocal calls or use of echolocation signals can be interpreted as both presence/absence, i.e., the animals have moved away or a change in acoustic behaviours, for example, not vocalizing as before. Active sonar systems can offer an attractive alternative in behavioural response studies potentially providing high-resolution spatial and temporal tracking of non-vocalizing or currently "quite" species. The main frequency of the sonar signal chosen to be outside the hearing range of the species of interests. The potential, however, exists for signal generation away from the design frequency of the sonar based on both the transducer/electronics as well as the pulsed nature of the signal. These signals might be perceived by the species of interest and therefore alter the behavioural response under study. Typical active sonar signals are analysed and compared to known hearing and integration periods for a range of species and the potential biological consequences of this out-of-band energy evaluated.

2:10

4pAB3. The use of active and passive acoustics for early detection and control of invasions. Francis Juanes (Biology, Univ. Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, juanes@uvic.ca)

As the threat of global invasions increase it is critically important to develop tools for early detection and control of invasive species. The earlier invasives can be detected the more feasible their control and potential eradication will be. Surveys involve the use of sensors to find and verify the identity of new invaders quickly and efficiently. Here, I summarize what features of acoustic sensors make effective tools for early detection of invasives and present examples of their use in both terrestrial and aquatic environments. Acoustic sensors combined with visual sensors allow for species identification and localization when sounds are unknown or undescribed. Technology for use of passive acoustic sensors for early detection of invasives is still in its infancy, but use will expand as methodology for automatic detection and identification develops particularly in remote settings. Once an invasive species is detected, active acoustic methods can be used to prevent range expansion or facilitate biocontrol. I will present various examples of such control methods from studies targeting fish, bird, mammals and amphibians.

2:25

4pAB4. Vertical line array tilt revealed through snapping shrimp noise.
Edward Richards, Zhuqing Yuan, Hee-Chun Song, and William S. Hodgkiss (Scripps Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, edwardlrichards@ucsd.edu)

The shallow-water acoustic variability experiment (SAVEX15), conducted in the East China Sea, unexpectedly recorded a great deal of snapping shrimp noise on a 16-element (56-m length) vertical line array deployed in 100-m water depths. These impulsive events can be used to find the tilt of the moored array which was caused by the strong ocean currents in the area. In this environment, the recorded snaps of bottom-dwelling shrimp within a radius of more than 500 m were largely separated from one another in time, and coherent time domain beamforming was successful in localizing snaps in time and range. The sparse and impulse nature of the snaps allowed for a simple normalization and gave tens of snap detections per second after a threshold detector. The results from the automated detector showed it was necessary to search with the beamformer over a three-dimensional space for reliable performance: (1) arrival time, (2) source range, and (3) array tilt. The results of the beamformer search in array tilt space are consistent with independent acoustic tilt measurements of the same array, showing that snapping shrimp noise in this region can provide valuable inference for the acoustic environment.

2:40–2:55 Break

Invited Paper

2:55

4pAB5. Relationship between cetacean presence, their prey, and oceanographic conditions at an offshore pelagic environment.
Simone Baumann-Pickering, Jennifer S. Trickey, Ally Rice, Ana Širović (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Josiah Renfree, and David A. Demer (Southwest Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, La Jolla, CA)

Large mobile predators take advantage of offshore pelagic environments that tend to be surveyed at a low temporal scale. We present results from a one-year observation of passive and active acoustics, as well as oceanographic measures, taken at a moored location 150 nmi offshore southern California in 4000 m deep pelagic waters. The passive acoustic data shows presence of delphinids and baleen, sperm, and beaked whales throughout the recording period with species-specific seasonal occurrence. Variability in small pelagics, krill, and diel vertical migrators is evident from the active acoustic data. Multiple upwelling events and associated changes in dissolved oxygen at 500 m water depth, as well as in surface waters, were also recorded over the year. We will present results on the relationships among oceanographic variability, presence of pelagic species, and top predator presence. This study will provide valuable information on ecosystem processes in meso- and epipelagic waters on a relatively small spatial scale but very fine temporal scale.

Contributed Papers

3:15

4pAB6. Predator-prey interactions in the Southern California Bight.
Ana Širović (Texas A&M Univ. Galveston, 9500 Gilman Dr. MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA), and Joseph Warren (Stony Brook Univ., Southampton, NY)

The Southern California Bight (SCB) is an important foraging area for a variety of cetacean species, including blue whales and beaked whales. However, the interactions of these species with their prey (krill and deep-water squid, respectively) can be very difficult to observe. Moorings combining passive and active acoustics were deployed at two locations in the SCB in two consecutive fall and winter seasons. A High-frequency Acoustic Recording Package sampling at 200 kHz recorded the full range of baleen and beaked whale signals. The active acoustic system on each mooring included two Simrad Wide Band Acoustic Transceivers (WBAT). A bottom-mounted WBAT had an upward looking 70 kHz transducer and the one at 300 m depth had upward looking 70 kHz and 200 kHz transducers. Over the course of four months, blue whale songs and social calls were recorded on both moorings. Beaked whale echolocation clicks were commonly detected on the mooring deployed offshore, with only rare beaked whale encounters on the coastal mooring. Backscatter data indicated overlap in the presence of cetacean predators and their prey. Scattering layer dynamics were dominated by regular diel migration, with differences in scatterer abundance between mooring sites, as well as on 2–5 day time scales.

3:30

4pAB7. Acoustic estimation of the biodiversity of fish and invertebrates.
Xavier Mouy (School of Earth and Oceans Sci., Univ. of Victoria, 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, Xavier.Mouy@jasco.com), Fabio Cabrera De Leo (Ocean Networks Canada, Victoria, BC, Canada), Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada), and Stan E. Dosso (School of Earth and Oceans Sci., Univ. of Victoria, Victoria, BC, Canada)

The increase in climate and anthropogenic stressors in the marine environment can lead to important losses in habitat for fishes and invertebrates. Consequently, monitoring marine biodiversity has become a critical task for ecologists. Traditional biodiversity measurements are costly and logistically challenging and there is an increasing need to develop new techniques that are more suitable for long-term and large-scale monitoring. The objective of this work is to assess the efficiency of active and passive acoustics to monitor the presence and diversity of fish and invertebrates. This work uses a multi-instrument platform, deployed on Ocean Networks Canada's VENUS cabled observatory in the Strait of Georgia (British Columbia, Canada), comprised of a high-definition video camera with a pair of LED lights, a dual-frequency imaging sonar and a hydrophone. Fish and invertebrates are counted and identified using the data from the video camera and sonar. Several acoustic indices such as acoustic complexity indices and similarity sound clusters are computed from the hydrophone data. The time series of these indices estimated from the passive acoustic data are then compared to the camera and sonar recordings to assess the ability of passive acoustics alone to determine the presence and diversity of fish and invertebrates.

3:45

4pAB8. Continuous monitoring and classification of odontocete echolocation clicks from an ocean observatory. Samuel S. Urmy, Kelly J. Benoit-Bird, John P. Ryan, and Danelle E. Cline (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, urmy@mbari.org)

Passive acoustic monitoring is a valuable technique for detecting the presence and inferring the activities of odontocetes, but is often limited by power and data-storage constraints of self-contained recorders. Since July 2015, a broadband (10 Hz–128 kHz) hydrophone has been recording continuously at the Monterey Accelerated Research System (MARS), a cabled observatory in Monterey Bay, California, USA. This acoustic record is notable for its combination of duration, bandwidth, and completeness. An automated detector identified more than 200 million unique clicks in this dataset, and extracted a variety of time-domain, spectral, and cepstral features from each click. Depending on the criteria used, clustering algorithms identified 4-8 click classes consistent with those of local odontocete species, including dolphins and beaked whales. Clicking rates were highly variable, with a median of 0.2 clicks minute^{-1} and a mean of 165 ($\pm$ 130 standard error) clicks minute^{-1}. Echolocation activity was 1-2 orders of magnitude higher at night than during daytime, and was seasonally higher in fall and winter. Prior work has shown that mesopelagic sound-scattering layers in Monterey Bay are densest at these times of year, suggesting higher availability of prey. Continuous passive monitoring has great potential to improve our understanding of these species' foraging ecology, especially when integrated with environmental measurements and active acoustic measurements of their prey fields.

4:00–4:20 Panel Discussion

THURSDAY AFTERNOON, 8 NOVEMBER 2018 ESQUIMALT (VCC), 1:00 P.M. TO 3:30 P.M.

Session 4pAO

Acoustical Oceanography: Topics in Acoustic Oceanography

Boris Katsnelson, Chair
Marine Geosciences, University of Haifa, Mt. Carmel, Haifa 31905, Israel

Contributed Papers

1:00

4pAO1. Broadband acoustic observations of individual naturally occurring hydrate-coated bubbles in the Gulf of Mexico. Elizabeth F. Weidner (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu), Kevin Jerram (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), and Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH)

Methane has been observed escaping from the seafloor as free gas bubbles. During ascent through the watercolumn, these bubbles undergo several processes, including accumulation of surfactants on the bubble surface. Surfactants have been theorized to increase the longevity of bubbles, facilitating methane transport to the upper ocean. Surfactants take many forms, including biological detritus, oil coatings, and hydrate shells. In 2011 and 2012, hydrate shelled bubbles were visually observed by ROV Hercules in the Northern Gulf of Mexico at the Biloxi Dome. The NOAA Ship *Okeanos Explorer* returned to the Biloxi Dome in March of 2018 with a broadband (10 kHz bandwidth) echosounder with vertical range resolution of approximately 7 cm. Individual bubbles were acoustically observed, rising from the seafloor several hundred meters before exiting the echosounder beam. In addition, the *Okeanos Explorer* made multiple survey passes over the seep site, mapping the vertical rise of the bubble plume at almost 1000 m. The dataset supports detailed modeling of the effects of hydrate coatings on acoustic response and provides an opportunity to estimate the dissolution rate of naturally occurring hydrate-coated bubbles.

1:15

4pAO2. Observations of acoustic intensity fluctuations and ambient noise near the surf zone. Kaustubha Raghukumar (Marine Sci. and Eng., Integral Consulting, 200 Washington St., Ste. 201, Santa Cruz, CA 95060, kraghukumar@integral-corp.com) and Kara Scheu (Marine Sci. and Eng., Integral Consulting, Portland, OR)

The recently completed Inner Shelf Direct Research Initiative (IS-DRI) experiment examined in great detail the physical oceanographic processes involved in the exchange of heat and momentum from outside the surf zone to the inner continental shelf, with a focus on features such as rip currents, fronts, and nonlinear internal waves. During this experiment, 27 kHz acoustic transmissions were also conducted with the goal of understanding nearshore acoustic fluctuations and their impact on acoustic communications and sonar performance. Acoustic intensity fluctuations are examined during two week-long periods during which intensity fades greater than 20 dB were observed. Variability spectra were found to be dominated at lower frequencies by tidal oscillations and higher frequency variability is attributed to intense nonlinear internal wave activity. Ambient noise spectra were found to be anisotropic, with low frequency spectral levels dominated by noise from wave breaking. When significant wave heights exceeded 3 m, a significant drop in spectral levels is observed, likely due to attenuation of sound by bubble plumes from breaking waves which are then ejected offshore by rip currents. Images from airplane flight missions confirmed the presence of offshore bubble plumes during swell events.

1:30

4pAO3. Determining spectral properties of Bowhead whale vocalizations through automated methods. Thomas V. Caero (U.S. Coast Guard, 31 Mohegan Ave., Chase 7162, New London, CT 06320, vinson.caero@gmail.com), Aaron Thode, and Christopher M. Verlinden (Scripps Inst. of Oceanogr., La Jolla, CA)

Determining spectral properties of whale vocalizations is valuable to the study of marine mammal behavior. However, manually measuring these properties is tedious, which motivates the search for automated methods. Linear regression and Radon transform methods were developed to automatically measure the slope of Bowhead whale frequency-modulated (FM) sweeps from spectrogram data. The Radon transform method was further expanded to include classification of whale vocalizations to aid in further analysis. The data collected using these methods were used to study the acoustic dispersive nature of marine mammal vocalizations in an ocean waveguide. Possible relationships were explored between the change in slope of the whale vocalization and range between each animal and the receiver.

1:45

4pAO4. Estimation of gassy layer parameters in shallow water sediment using low frequency ship noise. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il) and Andrey Lunkov (A.M.Prokhorov General Phys. Inst., Moscow, Russian Federation)

Sound speed in gassy sediments can reach very low values (~100 m/s) which provides high reflection coefficient for not very low frequencies (above a few tens of Hz) and specific properties of frequency dependence in the sound reflection/propagation phenomena in shallow water which allows us to estimate parameters of gassy layer (thickness and gas concentration). In the present study, shipping noise is used for estimation of mentioned parameters in Lake Kinneret (Israel). The sound noise from R/V "Hermona" was recorded at a vertical line array, moored in the lake. R/V was moving toward VLA at the speed ~4 m/s along 37 m isobath. Fourier components of the noise in the band 22–76 Hz were identified and analyzed. Mode selection was implemented to determine the mode amplitude as a function of range to the source for up to 4 normal modes. Matching mode attenuation coefficients with modeled ones for several frequencies, the thickness of the gassy layer and bubble concentration were estimated. The obtained values (thickness ~40 cm and gas content ~0.25%) correlate well with the results of direct measurements using X ray CT for cores. [This work was supported by RFBR 16-32-60194.]

2:00

4pAO5. Ray identification and ranging issues in deep water. John Boyle (OASIS Inc., 5 Militia Dr., Lexington, MA 02421, oceanpattern@gmail.com) and Kevin D. Heaney (OASIS Inc., Fairfax Station, VA)

Ocean acoustic positioning systems require accurate measurements of travel time from several acoustic beacons. When receiver position is unknown, the uncertainties in the sound speed, source position, currents, internal waves, and tides must be addressed and bounded. High travel time accuracy has been achieved in the deep ocean in ocean acoustic tomography experiments via a combination of long-duration acoustic arrival time-series, accurate source/receiver positioning, and comprehensive, post-experiment ocean modeling and data assimilation. For offline autonomous underwater positioning, those are not available. Strategies to reduce acoustic forward model uncertainties and improve acoustic ranging in deep water will be discussed.

2:15

4pAO6. Target strength measurements of pointhead flounder, *Cleisthenes pinetorum*, a flatfish without a bladder. Tohru Mukai (Faculty of Fisheries Sci., Hokkaido Univ., 3-1-1 Minato, Hakodate, Hokkaido 0418611, Japan, mukai@fish.hokudai.ac.jp), Naizheng Yan (Graduate School of Fisheries Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Jun Yamamoto (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan), and Kohei Hasegawa (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Japan)

There are only a few acoustic surveys of the bladderless fish. As is known by all, Atlantic mackerel was surveyed these years by echo-sounder and showed the high target strength at high frequency. In northern Japan, an interesting fish attracted the attention these years which called pointhead flounder, *Cleisthenes pinetorum,* without a bladder. It is a flatfish but captures prey in the middle water column. And the body shape is not the normal spindle-shaped, but a flat shape, which may cause different acoustic characteristics with Atlantic mackerel. We did the surveys in Funka Bay, northern Japan and keep the samples to the tank to measure the acoustic characteristics of live flounder. The acoustic characteristics of pointhead flounder measured both using the tether method (38, 120 kHz) and free swimming method (120 kHz). During the measurements of free swimming method, we also observed the swimming actions of pointhead flounder and calculated the swimming angle at the same time. We also discussed the characteristics of TS of the swimming angle.

2:30

4pAO7. Effect of the wind-generated bubble layer on forward scattering from the ocean surface. Yuzhe Fan, Haisen Li, and Chao Xu (Harbin Eng. Univ., No. 145, Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, yuzhe.fan93@gmail.com)

Wave scattering from pressure release surfaces is a classical subject with a long history. The standard approach which treat the surface probabilistically has been well developed. Over the last decade, some problems are more effectively solved with the deterministic approach. However, the complexities of upper ocean affect acoustic scattering from the ocean surface, and wind-generated bubbles are one of the most important complexities. In this paper, the effect of wind-generated bubble layers on forward scattering from the ocean surface is studied. The extended Hall-Novarini model is used to describe the population spectral density of the bubble layer. The time-domain Helmholtz-Kirchhoff integral is used to describe the acoustic scattering from pressure release surfaces. The free-field Green's function for an inhomogeneous medium is formulated using ray method. The advantages of such conjunction are as follows: (1) the geometric shadowing effects generally neglected in the H-K integral can be handled by ray method; (2) the Helmholtz-Kirchhoff integral can provide a correct solution even in caustics zones. The numerical simulation reveals that, for large wind speeds, the amplitude of scattering signal is negligible in contrast to the amplitude of direct path signal. In such situation, the reverberation will be dominated by volume scattering of wind-generated bubbles.

2:45

4pAO8. Analysis of frontal eddies effect and horizontal refraction in Gulf of Mexico with Hybrid Coordinate Ocean Model (HYCOM). Yao Xiao (Inst. of Acoust., Chinese Acad. of Sci., 801 Ferst Dr., George W. Woodruff School of Mech. Eng., Atlanta, GA 30332, xyao62@gmail.com), Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

The cold and warm water carried by the eddies can significantly change the structure of the sound speed field, which is the key factor affecting long range deep-water sound propagation. Investigations of individual eddies have typically employed extensive deep bathy thermography readings. However, these procedures typically require a high density of readings throughout the oceanic region containing the eddy, requiring fairly extensive ship time and equipment. Further, results from these procedures are restricted to regions within the eddy where data were acquired. Fortunately, with the development of ocean models, it is convenient to obtain high-resolution ocean state estimates in global ocean as well as in regional ocean areas. In this presentation, we use the Hybrid Coordinate Ocean Model (HYCOM) extensively for high-resolution simulations to quantify the effects of vertical and horizontal refraction caused by a realistic ocean environment and actual bathymetry in Gulf of Mexico. For fixed source and receiver configurations, the impact of frequency, source depth in received pressure field and influence of frontal eddies on the travel time (and arrival angle) are examined. Moreover, the scenario and computational approach for a simulated long-range transmitted signal through a dynamic ocean with eddies is presented. Depth-averaged transmission loss, which computed by averaging the acoustic pressure-squared in the water over depth, is used to facilitate the observation of 3D horizontal refraction through frontal eddies.

3:00

4pAO9. The arrival time of mode one in a stochastic ocean. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

The travel time for end of the final finale is often used is in inversion algorithms for acoustic tomography experiments when there is impreciseness due to ship and mooring positions and/or motions. The rationale is the first mode has the maximum slowness and higher order ones must arrive earlier, so the finale must solely be composed of energy from the first mode. This places a constraint on the tomographic sections, e.g., on the summation of the ray path or mode group slownesses. In a two papers Dozier and Tappert (JASA, 1978) examined the re population of the mode space for signals propagating in a stochastic ocean described by a Garrett & Munk model for internal waves. In the limit of an energy conserving ocean, i.e. no loss through boundaries, the limit of the population goes to an equipartition population density which was verified by numerical experiments. For more realistic oceans with boundary losses, the limit is a race between loss and scattering. This complicates what can be well identified as mode one at long ranges at low SNR's. We perform a numerical experiment by spatial filtering for mode one along the range dependent path to select just its energy in the finale. Earlier NUMERICAL studies by the author, (JASA, 1998) suggests just five percent of the energy remained at one megameter ranges with a 1/2 Garrett/Munk ocean. [Work supported by ONR.]

3:15

4pAO10. Study on acoustic characteristics of pointhead flounder *Cleisthenes pinetorum* and juvenile walleye pollock *Gadus chalcogrammus*. Naizheng Yan (Graduate School of Fisheries Sci., Hokkaido Univ., 3-1-1, Minatocho, , Hakodate, Hokkaido 041-8611, Japan, ynz_1992@outlook.com), Tohru Mukai (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Hokkaido, Japan), Jun Yamamoto (Field Sci. Ctr. for Northern Biosphere, Hokkaido Univ., Hakodate, Japan), and Kohei Hasegawa (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Japan)

In this study, the acoustic differences between the pointhead flounder and the juvenile walleye pollock were examined using a quantitative echosounder around the Funka Bay, Japan and the acoustic characteristics of the pointhead flounder have been surveyed. Acoustics data of the fish were monitored using a Simrad EK60 (38, 120, 200 kHz) split-beam echo sounder in the field. The target strength (TS) and swimming angle of free swimming pointhead flounders were measured in a seawater tank (length: 10 m, width: 5 m, and height: 6 m). As the result, pointhead flounder schools presented a patch shaped echo on the echograms, whereas the distribution patterns of the juvenile walleye pollock schools were layered. The volume backscattering strength (SV) of the target schools extracted from the echograms showed that the pointhead flounder presented a higher SV at high frequency. In contrast, the juvenile walleye pollock showed higher SV at a low frequency. For pointhead flounder, the distribution of pitch angle was measured both by camera and echo-sounder at the experiments in the tank and shown the same distribution pattern. The TS of pointhead flounder is large bigger than other bladderless fish.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 — SIDNEY (VCC), 1:30 P.M. TO 3:45 P.M.

Session 4pBA

Biomedical Acoustics: Biomedical Acoustics II

Jeffrey A. Ketterling, Chair
Riverside Research, 156 William St., New York, NY 10038

Contributed Papers

1:30

4pBA1. Focusing ultrasound into the kidney using 3D patient models. Magda Abbas, Constantin Coussios, and Robin Cleveland (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, magda.abbas@eng.ox.ac.uk)

The incidence of kidney cancer worldwide is on the rise particularly in older patients. The current gold standard of treatment is surgical resection, which has significant morbidity. HIFU is an attractive candidate for treatment of kidney tumours, as there is acoustic access and it has fewer risks then resection. However, clinical trials have had disappointing outcomes with intervening layers of fat identified as a significant obstacle. Here CTs from three patients treated in a clinical trial were segmented into different tissue types. Then acoustic simulations were undertaken using the k-Wave Matlab toolbox for a single element 0.95 MHz transducer as was used in clinical trials. Similar to previous reports we find that refraction due to fat layers results in fragmentation of the focal volume, in some but not all patients. The fragmentation was found to be sensitive to transducer placement. Therefore, a planning step was introduced where a virtual source was placed at the target location and then the predicted acoustic field outside the body was used to determine the best position for the transducer based on the received phase and amplitude. After optimal placement of the transducer there was up to 25% increase in the focal gain.

1:45

4pBA2. Ultrasonic bioreactor to monitor 3D culture human engineered cartilage. Juan M. Melchor (Structural Mech., Univ. of Granada, Politecnico de fuentenueva, Granada, Granada 18071, Spain, jmelchor@ugr.es), Elena Lopez-Ruiz (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Granada, Spain), Juan Soto (Dept. of Optics, Faculty of Physical Sci., Complutense Univ. of Madrid, Madrid, Spain), Gema Jimenez, Cristina Antich (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Granada, Spain), Jose Manuel Baena (Breca Health Care, Granada, Spain), Macarena Perán (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Jaén, Spain), Guillermo Rus (Structural Mech., Univ. of Granada, Granada, Spain), and Juan Antonio Marchal (BioPathol. and Regenerative Medicine Inst. (IBIMER), Ctr. for Biomedical Res., Granada, Spain)

Engineered cartilage tissue is one of the most promising treatments for articular cartilage pathologies. In this work, an ultrasonic bioreactor was designed to implement a non-invasive real-time monitoring of the neo-cartilage tissue formation processes through signal analysis. Polylactic acid (PLA) scaffolds were printed and seeded with human chondrocytes. Then, they were cultured in an ultrasound (US)-integrated bioreactor. The readings from the ultrasonic sensors were analyzed by numerical models of the ultrasound-tissue interaction and by a stochastic treatment to infer the extracellular matrix (ECM) evolution. To reconstruct the velocity and attenuation from the recorded signals, a genetic-algorithm based inverse problem (IP) was combined with an iterative computational propagation. The ultrasonic data were validated against evolution measurements of the *in vitro* 3D chondrocyte cultures assessed by proliferation and morphological observations, qualitative and quantitative biochemical parameters and gene expression analysis. The significant correlation shown between glycosaminoglycans (GAG) and collagen II (Col II) expression with the elastic damping evolution of the novo ECM ($R = 0.78$; $p < 0.001$) and ($R = 0.57$; $p < 0.01$), respectively, reinforces the feasibility of using ultrasound to evaluate chondrocyte functionality. Consequently, US can be used to monitor chondrocyte proliferation and ECM formation in the context of 3D cartilage engineering.

2:00

4pBA3. Analysis of a dual aperture approach and standing wave suppression pulse sequences for controlled transvertebral focused ultrasound delivery in *ex vivo* human thoracic vertebrae. Stecia-Marie P. Fletcher and Meaghan A. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. M7-302, Toronto, ON M4N 3M5, Canada, sfletcher@sri.utoronto.ca)

Focused ultrasound (FUS), in conjunction with microbubbles (MB), can open the blood-brain barrier to aid therapeutic delivery. Its feasibility for blood-spinal cord barrier opening (BSCBO) has been demonstrated in small animals. Controlled transvertebral delivery of FUS to the human vertebral canal requires low frequencies to penetrate thick vertebral bone. The resulting focal depth of field (DoF) is comparable to the size of the canal, promoting the formation of standing waves (SW) and potentially compromising treatment safety. Through k-Wave simulations and experimental acoustic field scans, a confocal, dual aperture approach was investigated in combination with SW suppressing pulses (linear chirp, short bursts and random phase shift keying (PSK)) to simultaneously reduce DoF and mitigate SWs. Two transducers (470/530 kHz, angle 90°) reduced the DoF by 85% compared to a single transducer (500 kHz). While all modified pulses reduced SWs relative to a 30 cycle sinusoid, short bursts performed significantly better than longer burst methods such as linear chirps ($p = 0.03$, (59 ± 15) vs (34 ± 11)% reduction). In combination with PSK, short bursts implemented in the dual aperture configuration mitigated SWs, while producing a uniform focal spot. The feasibility of using confocal, dual-aperture FUS to create a controlled focus within *ex vivo* vertebrae has been demonstrated. Next steps will include investigating MB emissions under short burst, PSK exposures, and characterizing spectral content associated with BSCBO and tissue damage.

2:15

4pBA4. Passive elastography monitoring of a high intensity ultrasound treatment: A study of feasibility in *in vitro* liver. Bruno Giammarinaro, Victor Barrère, David Melodelima, and Stefan Catheline (Univ. Lyon, Université Lyon 1, INSERM, LabTau, 151 cours Albert Thomas, Lyon Cedex 03 69424, France, bruno.giammarinaro@inserm.fr)

High intensity focused ultrasound (HIFU) is a non-invasive modality of intervention allowing thermal ablation in soft tissues by locally increasing temperature. Thermal lesions can be observed as a change in tissue elasticity properties, and so in shear wave velocity, by elastography. Most of studies based on ultrasound imaging used shear waves created by acoustic radiation force. Some of these studies presented elasticity measurements during a treatment in tissues like livers. They demonstrated that the elasticity increases during the treatment and so that the lesion becomes progressively harder than the original tissue. Acoustic radiation force method still presents some difficulties to obtain elasticity in deep tissues. However, in the human body, a natural noise due to cardiac activity or arterial pulsatility can be used to characterize the elasticity in using noise correlation techniques, it corresponds to passive elastography. This method depends on the imaging technique and so can image deep tissues. The objective is so to study the feasibility of using passive elastography technique during a treatment. Here, experiments are performed in *in vitro* porcine and bovine livers, heated with an acoustic transducer till 80°C and imaged with a high framerate ultrasound imaging device.

2:30–2:45 Break

2:45

4pBA5. Characterizing tissue calcifications using the color Doppler ultrasound twinkling artifact. Scott A. Zinck and Julianna Simon (Acoust., The Penn State Univ., 201E Appl. Sci. Bldg., University Park, PA 16802, 19scottz@gmail.com)

Rapid Doppler shifts highlight some kidney stones with a rapid change of color in ultrasound imaging in what is termed the "twinkling artifact." While many hypotheses exist to describe the origin of twinkling, the currently accepted hypothesis is that surface microbubbles are stabilized in the crevices on the surface of the kidney stone. The objective here is to evaluate the distribution of bubbles stabilized on the kidney stone surface and to determine whether other calcified tissues display the twinkling artifact. A Verasonics® research ultrasound system with the Verasonics® L22-14 and Philips/ATL L7-4 and P4-2 transducers was used to quantify twinkling on kidney stones and in minerals deposited by osteogenic stem cells over a range of frequencies. Preliminary results on kidney stones suggest that surface roughness and chemical composition influence the magnitude of twinkling. Osteogenic stem cells were also found to twinkle when minerals were present; correlating twinkling magnitude with the quantity and distribution of mineral deposition is a topic of ongoing research. The appearance of twinkling on even small tissue calcifications indicates ultrasound is very sensitive to mineral deposition and suggests that bubbles may be byproduct of the calcification process.

3:00

4pBA6. Stress reduction effects using deep breathing rhythm pattern based on speech utterance system. Seonggeon Bae (Div. of Comput. Media Information Eng., Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and myungjin bae (Information & TeleCommun., Soongsil Univ., Seoul, South Korea)

A breathing with a certain pattern plays an important role in heart breathing, and it is very important for daily life by relieving physical tension and finding psychological stability. In order to verify this, we analyzed the phonetically evoked organs and studied the appropriate method. In the

general study, a reliable effect of 83% was confirmed in 200 experimental groups. In the future, we will perform many studies from the viewpoint of speech signal processing using these features.

3:15

4pBA7. A study on voice enhancement using palm reflections. Hyung Woo Park and Myungjin bae (IT, Soongsil Univ., 1212 Hyungham Eng. Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, pphw@ssu.ac.kr)

Peoples communicates with others using voice that is essayist way. Voice is a means of conveying your opinions or thoughts. In complex social life, voices become rough and tone changes due to various causes such as condition, stress, singing, effect of drugs and changing the pitch due to changes in the numerical value of hormones. This will result in the other party not being able to communicate. In order to solve this treatment from the hospital. In this paper, we introduce the phenomenon of improving the clarity of human voice by utilizing the reflected sound waves of the palm. In previous research, We proposed the methods that using voice recorded or a solid surface mug. In this study, we propose method that peoples own voice reflected from the palm of their vocal organ. Speak the voice for "ah," "e," "o", "ooo" for a minute and make a sound wave in each mouth and massage it. The reflected voice from the palm stimulates delicate massage of the skin cells of the mouth, further enhancing the original voice enhancement function.

3:30

4pBA8. *Ex vivo* tissue preservation solutions for ultrasonic atomization. Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., Univ. 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), Yak-Nam Wang, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Ziyue Liu (Dept. of Biostatistics, Indiana Univ. Schools of Public Health and Medicine, Indianapolis, IN), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Tissue water content has been observed to influence ultrasonic atomization efficiency, which raises questions regarding *ex vivo* tissue storage time and viability. Here, we investigate the influence of tissue preservation solutions on acoustic properties and atomization efficiency. Samples from ten bovine livers were immediately submerged in phosphate-buffered saline (PBSaline), phosphate-buffered sucrose (PBSucrose), phosphate-buffered raffinose (PBRaffinose), or ViaSpan® clinical organ transplantation solution; no solution was also tested. Acoustic attenuation, sound speed, water content, and atomization efficiency were evaluated on either day 1 or day 2. For atomization efficiency, samples were partially submerged in water with the liver-air interface aligned with a 2-MHz focused transducer operating with 10-ms pulses at 1 Hz and peak pressures of 65/-16 MPa. Overall, no difference in atomization efficiency was observed. Water content in tissue was 66–78%, with PBSaline showing a significant increase between days ($p = 0.0013$). Only tissues in no solution showed a significant difference between days in acoustic attenuation ($p = 0.0003$). Sound speed measurements had more variation with only ViaSpan and PBSucrose showing no significant differences. As ViaSpan and PBSurcrose showed the most similarities in the tested acoustic properties, PBSurcose may be a low-cost alternative to the clinical ViaSpan solution for preserving *ex vivo* tissues. [Work supported by NIH DK043881 and EB017857.]

THURSDAY AFTERNOON, 8 NOVEMBER 2018 CRYSTAL BALLROOM (FE), 1:30 P.M. TO 3:55 P.M.

Session 4pMU

Musical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Computational and Experimental Investigations of Flow in Musical Instruments

Whitney L. Coyle, Chair
Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Chair's Introduction—1:30

Invited Papers

1:35

4pMU1. Speckle imaging of air flow. Thomas Moore and Whitney L. Coyle (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

Imaging airflow in ambient air is difficult, and it is especially difficult if the flow covers a large area. Schlieren photography is often used to image airflow, but it requires that the optics be at least as large as the area of observation, often making the cost prohibitive. Particle image velocimetry can also be used to visualize flow, but it requires specialized equipment, is computationally intensive, and imaging large areas can be challenging. We present an alternative method for imaging flow that is a variation of electronic speckle pattern interferometry. The system is easy to construct, can image large areas, and can be used to visualize both transient and steady-state flow.

1:55

4pMU2. Using speckle imaging of air flow inside musical instruments to improve computational models. Whitney L. Coyle and Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, wcoyle@rollins.edu)

Measuring air flow inside a clarinet mouthpiece is a difficult task but an important characterization step if the computational model of the instrument is to be as accurate as possible. Several studies have focused on solving this problem computationally, and some experimentally, using Schlieren imaging techniques. Though effective, the Schlieren technique is tricky and time consuming and does not offer the spatial resolution necessary to detect the small changes in flow. For the clarinet, there are two flow contributions in the clarinet mouthpiece (AC/DC flow and reed-induced flow) that need to be better understood experimentally in order to include them accurately in the computational models. Here, this flow characterization begins with a simpler, non-moving-reed instrument—an organ pipe. Building off of the previous presentation, this work details the physical measurements possible with the ESPI technique and will briefly discuss the implications of this discovery on one particular computational model.

2:15

4pMU3. Navier-Stokes-based computational studies of a lip reed instrument. Nicholas Giordano (Phys., College of Sci. and Mathematics, Auburn Univ., Auburn, AL 36849, njg0003@auburn.edu)

Direct numerical solutions of the Navier-Stokes equations have been used to calculate the behavior of the flow velocity and density in a trumpet-like instrument [1]. An essential ingredient of this study is a dynamical model of the player's lips, which are described by the Adachi-Sato model [2] and incorporated into the numerical treatment using the immersed boundary method. The results are used to test previous models which rely on the Bernoulli approximation to estimate the pressure near the lips. That approximation is found to fail badly under the conditions commonly found in these instruments. Our numerical method can be readily extended to other wind instruments such as the clarinet. Work supported by NSF grant PHY1513273. [1] N. Giordano, "Physical modeling of a conical lip reed instrument," JASA 143, 38–50 (2018), to be published. [2] S. Adachi and M. Sato, "Trumpet sound simulation using a two-dimensional lip vibration model," JASA 99, 1200–1209 (1996).

Contributed Papers

2:35

4pMU4. Using Navier-Stokes modeling to design a better recorder. Jared W. Thacker and N. Giordano (Phys., Auburn Univ., 210 East Thach Ave., Apt. 24E, Auburn, AL 36830, jwt0024@tigermail.auburn.edu)

We use the Navier-Stokes equations to model how air flows in the mouthpiece region of the recorder. Different geometrical designs for this region are investigated, and the effect of these modifications on the tonal properties are studied. We find that simple changes from the standard mouthpiece geometry can have large effects on the resulting spectra which we have traced to changes in the vortex patterns near the labium. Our modeling results have been tested by using 3D printing to make instruments suggested to be most promising. The modeling results also reveal how subtle changes in features like the sharpness of the labium can have a profound effect on the behavior. [Work supported by NSF grant PHY1513273.]

2:50–3:10 Break

3:10

4pMU5. Experimental investigation of acoustic radiation from a flute with an artificial blowing device. Ai Natsubori (Dept. of Mech. Eng., Toyohashi Univ. of Technol., 1-1 Hibarigaoka,Tempaku-cho, Toyohash-shi, Aichi-ken 4418580, Japan, natsubori@aero.me.tut.ac.jp), Hiroshi Yokoyama (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi,Aichi, Japan), Akiyoshi Iida (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi-shi, Aichi-ken, Japan), and Keita Arimoto (Res. & Development Div., YAMAHA Corp., Hamamatsu-shi, Shizuoka-ken, Japan)

To clarify the effects of air jet conditions controlled by flutists on acoustic radiation, experiments were performed by an artificial blowing device with an artificial oral cavity. The experimental parameters are the flow rate, the jet angle between the jet issuing from the cavity exit and the surface of the mouth hole, the jet offset to the edge and the distance from the cavity exit to the edge. Each parameter was changed independently. The reference values for the conditions were determined by referring to the actual playing conditions, of which jet angle was measured by the Schlieren method in this research. The Reynolds number based on the cross-sectional mean velocity and jet height was changed from 1700 to 2560 by adjusting the flow rate. In this range, as the flow rate was increased, the fundamental frequency became higher, keeping the first acoustic mode. The fundamental frequency became lower for the larger jet angle within the intense radiation. When the cavity exit was biased outside the edge by the jet height, the second harmonic became more intense while the fundamental frequency was approximately constant. The shift of the fundamental frequency to the higher mode was observed for a longer jet distance.

3:25

4pMU6. Direct aeroacoustic simulation of acoustic radiation in recorders with different windway geometries. Kimie Onogi (Dept. of Mech. Eng., Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Tempaku-cho, Toyohashi-shi, Aichi-ken 441-8580, Japan, onogi@aero.me.tut.ac.jp), Hiroshi Yokoyama (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi, Aichi, Japan), Akiyoshi Iida (Dept. of Mech. Eng., Toyohashi Univ. of Technol., Toyohashi-shi, Aichi-ken, Japan), and Tetsuro Shoji (Res. & Development Div., YAMAHA Corp., Hamamatsu-shi, Shizuoka-ken, Japan)

To clarify the effects of the windway geometry on the aeroacoustic feedback in the jet fluctuations in recorders, direct aeroacoustic simulations were performed along with experiments. The simulations were based on the compressible Navier-Stokes equations to predict the fluid-acoustic interactions. The volume penalization method was used to reproduce the flow and acoustic fields around the complex shape of the recorders. Two recorders with straight and arch-shaped windway were explored. The occurrence of mode change was observed at the higher velocity for an arch-shaped windway model compared with the straight windway model. The modified formulation of the negative displacement model (N.H.Fletcher *et al.*, 1976, J. Acoust.) was proposed based on the predicted jet fluctuations, where the jet fluctuations were divided into hydrodynamic and acoustic components. The ratio of the hydrodynamic component to the acoustic component near the windway exit was lower in the arch-shaped windway model than that in the straight windway model, whereas the amplification factor of the jet fluctuations was larger in the arch-shaped windway model. The relevance of these results and the mode change along the jet velocity is to be discussed.

3:40

4pMU7. Violin surface velocities and radiativity. Chironjeev Kanjilal and Chris Waltham (Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada, chironjeev.kanjilal@gmail.com)

Complete surface velocity maps and radiativity measurements have been made of several violins of varying quality. The instruments were excited laterally at the top of the bridge with a small impact hammer, giving a strong signal up to 4 kHz. The wood velocities were measured with a 0.2 g accelerometer placed sequentially on a grid, all over the surface. The grid was small enough to allow a detailed look at the bridge island. The f-hole velocities were measured with a 6 mm diameter microphone at nine positions per hole, and the acoustic pressures were converted to velocities with a propagating wavefront model. The surface measurements were compared to radiation measurements made with a 92 cm radius microphone array in an anechoic chamber. The surface measurements could be completed in a day, the radiativities in two hours. The resulting maps of operating deflection shapes, calibrated in m/s per N, allow a detailed examination of the vibrational behavior of each instrument, at frequencies up to and including the first torsional mode of the bridge.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 — SALON C (VCC), 1:00 P.M. TO 2:15 P.M.

Session 4pNSa

Noise, Speech Communication, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance II

Joonhee Lee, Cochair
Department of Building, Civil and Environmental Engineering, Concordia University, EV 6.231, 1515 Rue Sainte-Catherine O, Montreal, QC H3H1Y3, Canada

Z. Ellen Peng, Cochair
Waisman Center, University of Wisconsin-Madison, 1500 Highland Avenue, Madison, WI 53711

Contributed Papers

1:00

4pNSa1. Comparing resonance frequencies in muscle tissue before and after exercise induced damage. Garrett Jones, Cameron Smallwood (Mechanical Eng., Brigham Young Univ., CTB 435Q, Provo, UT 84602, garrett-jones.co@gmail.com), Brent Feland (Exercise Sci., Brigham Young Univ., Provo, UT), and Jonathan Blotter (Mechancial Eng., Brigham Young Univ., Provo, UT)

There are various approaches to measuring muscle fatigue, including such simple measures as surveys for qualitative data, and measuring power output for quantitative data. Electromyography is consistently used to determine muscle fatigue. However, a quantitative measure of muscle recovery after exercise-induced damage is yet to be defined. The objective of this study is to measure the resonance frequency of a muscle group using a 3D laser vibrometer system and make a correlation with both elastography stiffness measurements, and with recovery of a muscle group based on standard muscle fatigue measurement techniques. Measurement of the resonance frequency of a muscle group has been successfully performed, and there is good reason to believe that the resonance frequency of the muscle group does shift after exercise-induced damage occurs. The main outcome of this study will determine whether the shift in resonance frequency of a muscle group can be used to track recovery after a damage protocol.

1:15

4pNSa2. Workplace noise assessments by open-plan office occupants: Relationship with ISO 3382-3 metrics, and psychoacoustic parameters. Manuj Yadav, Densil Cabrera, James Love, Jungsoo Kim, and Richard de Dear (School of Architecture, Design and Planning, The Univ. of Sydney, Rm. 589, Wilkinson Bldg., 140, City Rd., NSW 2006, Australia, manuj.yadav@sydney.edu.au)

ISO 3382-3:2012 is the international standard for measuring the acoustics of open-plan offices. These measurements are performed in *unoccupied* offices, and it has recently been shown that some of the ISO 3382-3 metrics, especially distraction distance, are useful in predicting the occupants' perceived disturbance due to noise. The current research compares the ISO 3382-3 metrics, with several psychoacoustic parameters (loudness, fluctuation strength, etc.) derived from measurements done in an *occupied* state of open-plan offices, in terms of their usefulness in predicting occupants' perception of several aspects of workplace noise. This comparison was based on using measurements performed in several offices in both occupied and unoccupied states, along with a workplace noise survey that was conducted in these offices. The results will enable a more comprehensive understanding of noise-related issues in open-plan offices, as it involves both physical acoustic, and psychoacoustic considerations.

1:30

4pNSa3. A hearing-care toolkit for young musicians: Combined use of a noise dose measurement app and acoustical manekin. Romain Dumoulin (Ctr. for Interdisciplinary Res. in Music Media and Technol., CIRMMT, 527 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, romain.dumoulin@mcgill.ca), Francesco Tordini (Ctr. for Interdisciplinary Res. in Music Media and Technol., Montréal, QC, Canada), Jeremie Voix (École de technologie supérieure, Université du Québec, Montréal, QC, Canada), and Isabelle Cossette (Ctr. for Interdisciplinary Res. in Music Media and Technol., Montréal, QC, Canada)

Within a joint initiative with the Schulich School of Music at McGill University, the authors developed an original approach to raise awareness about hearing health by combining a participative assessment with a novel noise exposure risk indicator. The overall noise exposure is estimated using two complementary measurement systems addressing, respectively, the contribution determined by *(i)* the use of portable music players and *(ii)* music and non-music activities. The first system is a "listening-level measuring kiosk," a freely accessible station on which students can place their own headphones or earphones and playback (at their preferred listening levels) music excerpts from their own playlist on an acoustical manikin. The second system is a personal, smartphone-based, "noise exposure" portable measurement device. Users start with an initial training and an audiometry screening, followed by the collection of exposure data during a 4-week assessment period. Personalized results are provided based on individual exposure profile and initial audiogram. The validation of the design of a novel noise exposure risk indicator is based on the feedback from a focus group of young musicians. This paper describes the measurement systems, assessment process, and preliminary results from the ongoing pilot studies.

1:45

4pNSa4. Inciting our children to turn their music down: The AYE concept. Jeremie Voix and Romain Dumoulin (École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

According to the World Health Organization (WHO), more than 1.1 billion people are currently at risk of losing their hearing due to excessive exposure to noise. Of this, a significant proportion consists of children, youth and young adults who are exposing themselves to excessive levels of sound through various leisure activities (music players, concerts, movies at the theatre, dance clubs, etc.). To address this issue, many approaches have been developed, ranging from general awareness messages to volume limiters on personal music players. Nevertheless, it seems that, as with all dangerous behaviors, the most promising approach is a direct one in which a personalized assessment of the risk has been made and the person is made aware of the associated detrimental consequences. The present paper describes a counseling approach whereby the sound exposure is associated with an increased loss of hearing sensitivity. The proposed "Age of Your Ears (AYE)" metric is computed using the ISO 1999 multiregression model and predicts for a given age and a given sound exposure the resulting accelerated aging of the person's hearing. Preliminary results from pilot studies and focus groups with youths will be presented together with the underlying mathematical and statistical foundations.

2:00

4pNSa5. Debunking unusual false noise damage claims. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

We live in a society where noise seems to be everywhere and some of it is relatively loud, very annoying or even painful. In some cases we use noise to mask out noise, or we use ear plugs or we create quite spaces or "sound proof" rooms, just to get some peace and quiet. Nowadays, nearly everyone is aware of the annoyance or potential harm caused by other people's noise. This has been a liability issue for employer's and insurance companies. Laws, ordinances, regulations and standards (LORS) have been passed or otherwise put into place in an attempt to remedy excessive noise situations. Occupational hearing loss claims, violation fines, jail time and civil lawsuits have all taken place because of noise conflicts. This paper presents three examples where litigation was attempted for financial gain or to impose a heavy financial burden and force a business closure, when the plaintiffs enacted noise damage legal claims. Examples of noise from (1) a pick-up mounted railroad engine horn, (2) a very large rural private gun club with SWAT, law enforcement and military "Afghanistan-type" progressive urban and desert conflict scenarios, and (3) an EMT training class explosion simulation. Claims and outcomes are discussed.

Session 4pNSb

Noise and Psychological and Physiological Acoustics: Soundscapes

Kerrie G. Standlee, Chair
Daly-Standlee & Associates, Inc., 4900 SW Griffith Drive, Suite 205, Beaverton, OR 97005

Chair's Introduction—2:45

Contributed Papers

2:50

4pNSb1. Reducing the noise of tramways in urban areas. Rene Weinandy and Percy Appel (Noise Abatement in Transport, German Environment Agency, Woerlitzer Platz 1, Dessau-Rosslau 06844, Germany, rene.weinandy@uba.de)

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 the city 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 for city planners, engineers and politicians to make our cities quieter. The research project "Reducing the noise of tramways in urban areas"—initiated and supervised by the German Environment Agency—investigates potential noise abatement technologies and its implementation for tramways in Germany. Urban areas are growing worldwide and due to the resulting progressive consolidation, public transport including tramways will expand rapidly and subsequently raise severe environmental problems, i.e., noise. In order to protect the people, it is important to operate public transport as quiet as possible. However, the current legal, technical and economic conditions in Germany are only partially suitable to achieve that. Therefore, the research project investigates technical and operational concepts for noise abatement of tramways, possibilities of constructional sound insulation and legal aspects for the implementation of noise abatement measures. The presentation identifies and describes obstacles for the market penetration, operational problems, maintenance, and legislation.

3:05

4pNSb2. Measurement of the sound directivity of tonal and broadband reversing alarms in various controlled environments. Olivier Robin, Tamara Krpic (Groupe d'Acoustique de l'Université de Sherbrooke, Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Olivier.Robin@USherbrooke.ca), Hugues Nélisse (Institut de recherche Robert-Sauvé en santé et en Sécurité du travail (IRSST), Montreal, QC, Canada), and Alain Berry (Groupe d'Acoustique de l'Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC, Canada)

Tonal reversing alarms are a recognized source of noise annoyance and recent broadband alarms target a reduction of the environmental impact of backup sound alarms. From the worker and safety point, the sound field generated behind vehicles by broadband alarms is considered more uniform and its source easier to localize in space, a main drawback of tonal alarms being to exhibit large amplitude variations in the produced sound field pattern. Very few datasets are nevertheless available in order to objectively compare these two alarm types. The zone where the generated sound field is studied is also almost all the time restricted to the rear of the vehicle, even though additional diffraction effects can occur around the vehicle due to its finite volume and geometry. For two commercial alarms (tonal and broadband), 3D directivity patterns are obtained from microphone array measurements performed in controlled conditions, i.e., in a hemi-anechoic room and a reverberant room. The backup alarms are either placed on a stand at the center of the room, or on a parallelepiped surface representing the rear part of a small fork lift. The effect of the floor condition, i.e., reflective or absorbing, is also studied in the hemi-anechoic room.

3:20

4pNSb3. Derivation of network-wide surface condition corrections for rail noise modelling. Briony Croft (SLR Consulting, 200-1620 West 8th Ave., Vancouver, BC V6J1V4, Canada, bcroft@slrconsulting.com), Tasia Balding (Translink, Vancouver, BC, Canada), Pascal Everton, and Jasen Stein (SLR Consulting, Calgary, AB, Canada)

Noise modelling and mapping techniques are increasingly being used to understand noise impacts to communities across large geographic areas. Rail transit noise around a network can vary considerably with factors such as track type, train speed, rail roughness/surface condition, among others. Reference source levels for a particular rolling stock type are normally derived by measurement at a particular location or locations, and applied around the network. When modelling noise network wide, it is important to understand how variable rail roughness can affect predicted noise levels at locations other than those used for measurements. With elevated guideway, it can also be difficult to access appropriate external measurement locations. This study describes a study undertaken for the Vancouver SkyTrain network using the Nord2000 rail noise prediction algorithm. Variations in rolling noise level of the order of 15 dB were identified and attributed to rail surface condition. Noise measurements inside a test train were used to determine frequency dependent train speed coefficients and corrections to apply for rail condition around the network. This paper describes the approach used and the outcomes in relation to this city-wide rail noise mapping project.

3:35

4pNSb4. Sound quality metrics applied to road noise evaluation. Karolina Marciniuk (Multimedia Systems Dept., Gdansk Univ. of Technol., Faculty of ETI, Gdansk, Poland) and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

Road noise monitoring systems typically measure sound levels in specific time periods. The more insightful approach suggests to measure also the nature of noise. Sound quality of sounds such as car noise can be objectively evaluated by several parameters. One of them is psychoacoustic annoyance, described by loudness, tone color, and the temporal structure of sound. In this paper the assessment of several sound quality parameters, such as: loudness of time-varying noise, sharpness, fluctuation strength, and roughness, as well some additional parameters borrowed from the Music Information Retrieval area are presented. Then a comparison between parameter values obtained by means of the professional measurement system and a tool working in the Matlab environment is performed. The conducted investigations are carried out using recorded samples of an individual vehicle

pass-by in close proximity of the road, organized as a database of road traffic noise recordings. [This research was partly supported by the Polish National Centre for Research and Development within the Grant No. OT4-4B/AGH-PG-WSTKT.]

3:50

4pNSb5. Soundscape design connecting people with the environment. Taiko Shono (Office Shono, 4-32-6-614, Nishishinjuku, Tokyo, Shinjuku-ku 160-0023, Japan, shsh@aaa.email.ne.jp) and Hidemaro Shimoda (Acoust. Planning Corp., Tokyo, Minato-ku, Japan)

We have been engaged in soundscape design that enables people to connect with the environment and, in time, with the world beyond our immediate surrounds through sounds. We create sounds employing the elements on site: waves, wind, rain, spring water, etc., and strive to bring people into creative contact with our earth's environment and make people aware of something which may be present yet typically goes unnoticed. All of these works have been designed not as temporary performances but as permanent constructions responding to their environments. We have also dealt with "sound as media," employing not only audio from the natural world but also the familiar sounds of daily life, communities, and events, creating opportunities for people to rediscover the world around them in a new light. In addition, We have offered programs designed to awaken listening sense, including workshops at museums and other cultural facilities and university classes, with themes such as "aural adventure." In this paper, we show some examples of our activities and how we design the mechanisms of gathering or creating sounds.

4:05

4pNSb6. Urban soundscape: A case study in Taipei Dome. ChiaYuan Chuo and Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., No. 43, Keelung Rd., Sec.4, Da'an Dist., Taipei City, Taipei 10607, Taiwan, isnoopy307@gmail.com)

Taipei Dome (TD) is under construction and located in the heart of Taipei City. It is surrounded by prominent shopping district and popular cultural centers. The capacity of TD is 40,575 seats. Songshan Cultural and Creative Park is within the same block as TD and has retained the remaining green area within the block. Once it is in operation, the impact of the subsequent vehicular and crowd noise to the nearby buildings and green area is inevitable. To understand the nearby soundscape of TD, an itinerary with 8 locations were used. In-situ visual/aural observation with video recording were performed in 4 different time periods during one weekday and one weekend. A list of taxonomy of sounds was derived. While vehicular noise is the dominant sound category, natural sound such as bird chirping and crickets calling were also identified. Traffic counts were conducted. Scooters and cars were the two main vehicular types and with peak volumes during weekend evenings. Sound pressure level of each location was measured. Maximum peak sound pressure level of car noise reached as high as 109.5 dB. Mitigating expected vehicular noise and enhancing natural sound around TD should be considered by its design team.

4:20

4pNSb7. Soundscape approach integrating noise mapping in Namsan Promenade. Jisu Yoo (Energy Environment System Eng., Univ. of Seoul, 309 2nd Eng. Bldg., 163 Seoulsiripdae-ro, Seoul, Dongdaemun-gu 02504, South Korea, luvxjisu@gmail.com), Kyong Seok Ki (Sangji Univ., Seoul, South Korea), Hunjae Ryu (Energy Environment System Eng., Univ. of Seoul, Seoul, South Korea), Ji Suk Kim (Jungbu Parks & Landscape Management Office, Seoul Metropolitan Government, Seoul, South Korea), and Seo I. Chang (Environment Eng., Univ. of Seoul, Seoul, South Korea)

Namsan is a 262 m high mountain located in the center of Seoul. Its easy accessibility and tourist attractions such as a cable car and a tall tower on the top allure native and foreign people. It functions as a park rather than a mountain and has a 7.5 km promenade around it with a moderate slope where even the old and the weak enjoy walking. Some parts of the promenade are surrounded by trees and flowing waters but other parts are influenced by shuttle buses, roundabout road-traffic and various machines etc. Noise maps for the sources were generated to find the spatial distribution of the noise over the mountain including the promenade. By observation along the promenade, natural sound including mainly birdsong and water streaming sound were identified and sound maps were generated. To collect individual responses from 4 acousticians and 6 non-acoustic people about an on-site questionnaire of the sound environment along the promenade, a total of 37 spots were selected apart from each other by approximately 200 m. The responses were used to generate soundscape maps. Separate and combined analyses of the three acoustic maps were performed to propose some measures to improve sound quality of the promenade.

4:35

4pNSb8. Sound grade classification with sound mapping of national park trails in South Korea. Hunjae Ryu (Energy and Environ. System Eng., Univ. of Seoul, 2nd Eng. Bldg. 309, Seoulsiripdae-ro, Dongdaemun-gu, Seoul 02504, South Korea, pgryuno1@uos.ac.kr), Kyong Seok Ki (Horticulture and Landscape Architecture, Sangji Univ., Wonju-si, South Korea), Jisu Yoo, Seo I. Chang (Energy and Environ. System Eng., Univ. of Seoul, Seoul, South Korea), and Bo-Hyun Kim (Conservation Planning Div., Korea National Park Service, Wonju-si, South Korea)

Most of the national parks in South Korea are exposed to noise pollution caused by urban noise from adjacent cities. Especially, the road-traffic noise from the roads passing through or surrounding the parks has increased the exposure. In addition, the military and tele-communication facilities located in ecologically sensitive highlands are noise sources and affect nearby ecosystems. Therefore, it is important to preserve and maintain a comfortable sound environment to ensure the quality of the park trails for visitors and the habitat of animals and plants. The purpose of this study is to map a sound environment of national parks in South Korea and to classify sound grades of park trails. The park trails were categorized into 5 different groups based on acoustic factors such as noise level, loudness, sharpness, roughness, and tonality, and environmental factors such as land-use, ground coverage, vegetation type, and relative location which can be obtained from biotope map and GIS DB. This classification was based on factor analysis and cluster analysis. It is expected that this sound grade classification of the national park trails meets the right of visitors to know the sound environment, and that the managers of national parks can use it as a guideline to create a positive sound environment.

4:50

4pNSb9. A study on the hearing reverence psychology using rhythm patterns. SangHwi Jee, Myungjin Bae, and Myungsook Kim (Sori Sound Eng. Lab, Soongsil Univ. Seoul, Seoul 06978, South Korea, slayernights@ssu.ac.kr)

Humans are exposed to noise from the moment they are born until they die. Noise is a sound that humans do not like to hear. In the case of automobile horn noises, it produces a noise of more than 100 dB, which causes stress to people, which leads to reprisal driving and becomes a social problem. In the case of a light machine, it produces a very sharp sound, which causes stress when people listen. Therefore, we propose a new sound that moves the rhythm and pattern based on the existing sound. If we make rhythmic sounds using a rhythm rather than a constant rhythm, it induces less stress than conventional rhythmic sounds.

Session 4pPA

Physical Acoustics and Biomedical Acoustics: Interactions of Sound Beams with Objects II

Likun Zhang, Cochair
University of Mississippi, 145 Hill Drive, Oxford, MS 38677

Grant Eastland, Cochair
System Acceptance & Operational Readiness, Naval Undersea Warfare Center Division Keyport, 610 Dowell St., Keyport, WA 98345

Invited Papers

1:00

4pPA1. Strongly focused vortex beams by using flat Fresnel-spiral lenses. Noé Jiménez (Instituto de Instrumentación para Imagen Molecular (I3M), Consejo Superior de Investigaciones Científicas (CSIC), Edificio 8B, Acceso N, 1[a] Planta, Camino de Vera s/n, Valencia 46022, Spain, nojigon@upv.es), Vicente Romero-Garcìa (Laboratoire d'Acoustique de la Université du Mans UMR 6613, CNRS, Le Mans cedex 9, France), Lluis M. García-Raffi (IUMPA, Universitat Politècnica de València, València, Spain), Francisco Camarena (Instituto de Instrumentación para Imagen Molecular (I3M), Universitat Politècnica de València, Valencia, Spain), and Kestutis Staliunas (ICREA, Universitat Politècnica de Catalunya, Tarrasa, Spain)

We report geometrically-optimal diffraction gratings for sharp vortex beam focusing using Fresnel-spiral curves. The lenses are built based on the Fresnel-spiral, a spiral curve that combine the focusing properties of Fresnel zone plates and the phase dislocations produced by spiral gratings. On the one hand, the constructive and destructive interferences between open and opaque zones in the grating, in analogy to the Fresnel zone plate, allow sharp beam focusing. On the other hand, the spiral shape of the grating retains the helicity, rotating the phase of the diffracted waves and creating a phase dislocation along the axis. This allows the generation of geometrically optimal focused vortex beams, enhancing the field intensity at the focus up to 170 times. In particular, this system offers a tunable topological charge of the vortex beam by using different arms in the Fresnel-spiral diffraction grating, being the topological charge equal to the number of arms. Two different Fresnel-spiral diffraction gratings with topological charge of 1 and 5 are experimentally tested showing excellent agreement with theory and simulations. These diffraction gratings will allow the design of effective wave-matter interaction systems, with direct applications in industry and biomedical engineering.

1:20

4pPA2. Bessel beam superposition for analyzing off-axial scattering, forces, and torques. Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

Illumination of an arbitrary-order Bessel beam on an off-axial object is expanded as illuminations of a series of Bessel beam components of different orders on an axially centered object [L. Zhang, J. Acoust. Soc. Am. 143(5), 2796–2800 (2018)]. The expansion follows from a parallel-axis relation that is derived as a generization of Graf's addition theorem of Bessel functions. The off-axial Bessel beam expansion enables to understand the off-axial scattering, forces, and torques in terms of superposing that in the on-axial illuminations by the series of beam components. The superposition reveal negative axial radiation force and negative radiation torque in terms of contributions of different beam components in the on-axial configuration. The exact locations of the object for torque direction reversal are analytically identified for a small particle. The superposition also reveals that the off-axial radiation torque on a small particle exists even for higher-order vortex beams or vortices.

1:40

4pPA3. Expansion of linear focused sound fields using Bessel beams. Timothy D. Daniel (Phys., WSU, Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ivars P. Kirsteins (NUWCDIVNPT, Newport, RI), and Philip L. Marston (Phys., WSU, Pullman, WA)

Previous work on scattering by Bessel beams shows that expansion of incident sound fields in term of these beams has application to scattering [P. L. Marston, J. Acoust. Soc. Am. 122, 247–252 (2007)]. In this work, we expand a focused cylindrically symmetric sound field in terms of Bessel beam components. In particular, the authors are interested in a focused beam radiated from a spherical cap source. A physical optics model is applied to sound propagation close to the source to facilitate the calculation of the Bessel beam expansion coefficients. This type of model is useful for focused scattering [P. L. Marston and D. S. Langley, J. Acoust. Soc. Am. 73, 1464–1475 (1983)]. Once the expansion coefficients are found the sound field can be evaluated by the appropriate superposition. The model gives results in agreement with O'Neil [H. T. O'Neil, J. Acoust. Soc. Am. 21, 516–526 (1949)] and Chen [X. Chen, K. Q. Schwarz and K. J. Parker, J. Acoust. Soc. Am. 94, 2979–2991 (1993)]. Comparison is also made with direct integration of a Kirchhoff integral for both on and off axis pressures with good agreement. [Work supported by ONR.]

2:00

4pPA4. Experimental analysis of the instantaneous mechanical torque applied to objects by means of acoustic beams. Ruben D. Muelas-Hurtado (School of Civil Eng. and Geomatics, Universidad del Valle, Cali, Valle del Cauca, Colombia), Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Ciudad Universitaria Meléndez. Bldg. 351, Cali 760032, Colombia, joao.ealo@correounivalle.edu.co), and Karen Volke-Sepulveda (Instituto de Física, Universidad Nacional Autónoma de México - UNAM, Mexico City, Mexico D.F., Mexico)

In this work, we quantify the mechanical torque and the angular momentum transferred by an acoustic beam (AB) to disk-like samples of flat surface and chiral objects. Also, two types of acoustic beams are generated, i.e., quasi-planar and helical beams. To produce the AB, we have fabricated a multitransducer of 123 ultrasonic elements (40 kHz) deployed on a triangular lattice which allows us an electronic wavefront synthesis by modifying the phase of each emitter. The samples to be freely rotated are supported at its center of mass by a sharp tip of a thin metallic needle vertically located at the geometrical center of the multitransducer. The instantaneous angular response of samples of different size, made of cork and plastic, was extracted using a fixed video camera. The instantaneous torque exerted on each sample and the transferred angular momentum are estimated by fitting the output of a dynamic model of the sample behavior to empirical data. The experimental apparatus proves to be useful to estimate the acoustic torque induced by AB on objects and its dependency on the geometries of the samples and the AB wavefront, their relative size and the intensity of the acoustic beam.

2:20–2:35 Break

2:35

4pPA5. Acoustic radiation force manipulation of tissue, cell, and interactions with neurons. Hairong Zheng (Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., 1068 Xueyuan Ave., Shenzhen University Town, Shenzhen, China, hr.zheng@siat.ac.cn) and Feiyan Cai (Shenzhen Institutes of Adv. Technol.,Chinese Acad. of Sci., Shenzhen, China)

It is well known that an object in a sound field that absorbs, scatters, or reflects sound energy is subjected to the acoustic radiation force, and thus can be manipulated without contact. This unidirectional force has profound application in biomedicine. For example, the acoustic radiation force can be used to generate localized displacements in tissue, and the magnitude of the tissue displacement is inversely proportional to the local stiffness of the tissue. Based on this principle, the acoustic radiation force interaction with the tissue can be used for elasticity Imaging. The acoustic radiation force can also be used to manipulate cell's movement and simulate isolated neuron, which may have potential application in tissue engineering and ultrasound neuro-modulation. In this study, our recent research on the application of the acoustic radiation force for manipulation of tissue, cell and interactions with neurons, will be presented. New approaches or systems for generation acoustic radiation force with special application will also be shown.

2:55

4pPA6. Orbital motion of a particle levitated in a standing-vortex acoustical trap. Jhon F. Pazos-Ospina (Marco Fidel Suarez Military Aviation School (EMAVI), Colombia Air Force, Universidad del Valle, School of Mech. Eng. Edif. 351, Cali, Colombia, jhon.f.pazos@correounivalle.edu.co), Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Cali, Colombia), and Karen Volke-Sepulveda (Instituto de Física, Universidad Nacional Autónoma de México - UNAM, Mexico City, Mexico D.F., Mexico)

Stable acoustical tweezers and particle manipulation using vortex beams have recently been accomplished in air [Nature Commun. 6: 8661 (2015)] and in water [Phys. Rev. Lett. 116: 024301 (2016)]. In both configurations, a particle is trapped along the propagation axis of a focused vortex, where the acoustic field exhibits a pressure node. More recently, a stable orbital motion of a particle trapped off-axis was achieved by alternating between two counter-rotating vortices [*Phys. Rev. Lett.* 120: 044301 (2018)]. However, this kind of orbital motion had not been possible so far with a steady helical field having a single angular momentum state. In this work, we present the first experimental evidence of the simultaneous 3D trapping and stable orbiting of millimeter-sized particles in air due to angular momentum transfer in a standing wave trap created by the interference of two vortices. In contrast with previous work, the two vortices have the same helicity with respect to the laboratory reference frame, and thus the angular momentum of this field is well-defined. A description of the particle kinematics as a function of different control parameters is presented, along with a comparison between experimental measurements of the field and theoretical simulations.

3:15

4pPA7. Acoustic manipulation and actuation of bubbles in complex environments: Beyond the Bjerknes force . Diego Baresch (Dept. of Chemical Eng., Imperial College London, 4, Pl. Jussieu, Paris 75005, France, diego.baresch@upmc.fr) and Valeria Garbin (Dept. of Chemical Eng., Imperial College London, London, United Kingdom)

Micron-sized gas bubbles are notoriously difficult to isolate, handle and remotely control. Their large buoyancy in common liquids will usually force them to rise and burst at any gas/liquid interface or remain trapped against a solid boundary until dissolution. While bubble stability issues against dissolution have found numerous practical workarounds, the challenge remains at isolating and maneuvering a single bubble in free space to, for instance, perform precise single bubble dynamics experiments with applied ultrasound or to use them as active carriers for a specific payload deliverable on demand. Here we demonstrate that single-beam acoustical tweezers [D. Baresch *et al,* Phys. Rev. Lett., **116,** (2016)] can trap and manipulate in 3D a single bubble with the radiation pressure of helicoidal ultrasonic beams. Contrary to the situation where bubbles are trapped in the antinodes of a standing wave, the trapping vortex beam does not require oscillating volume changes of the bubble to generate a force, *i.e*., the trapping mechanisms cannot be explained in terms of Bjerknes forces. Viscous boundary layer effects and large bubble stability will be discussed. Opportunities for our manipulation technique to probe the high speed dynamics and rheology of particle-laden armored bubbles will also be presented [V. Poulichet *et al.,* PNAS, **112,** (2015)].

3:35

4pPA8. Three-dimensional acoustical radiation forces in the view of momentum conservation. Likun Zhang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

Analysis of acoustic radiation forces is essential for tailoring parameters for desired manipulations or for interpreting special force conditions. Previous theoretical derivation for acoustic radiation forces was from integration of stress tensor of the total fields. Given that the force is a second-order effect, interactions between the incident and scattered fields or between different beams contribute to the forces and complicate the analysis. Here the three-dimensional forces exerted by an arbitrary field is presented in the framework of momentum conservation where force superposition is possible for flexible analysis. The force is related to momentum transfers by ingoing and outgoing spherical waves and to momentum transfers associated with extinction and scattering. Superposition of forces based on orthogonality of momentum and insight into force dependence on beam and scattering parameters for special manipulations are addressed.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 SALON B (VCC), 1:00 P.M. TO 2:55 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Acoustics Outreach: Linking Physiology and Behavior for Future Collaborations II

Amanda Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins University School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Invited Papers

1:00

4pPP1. Insights from individual differences: Uncovering the code for frequency modulation. Kelly L. Whiteford, Heather A. Kreft, and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu)

Perceiving changes in frequency (frequency modulation; FM) and amplitude (amplitude modulation; AM) are essential for human and animal communication. Our goal was to utilize individual differences in sensitivity for FM and AM to better understand how FM is represented in the auditory system. Previous studies suggest that FM of low-frequency carriers (f_cs < ~4 kHz) at slow modulation rates (f_m < 10 Hz) is coded by precise, phase-locked spike times in the auditory nerve (time code). The same f_cs at faster rates may utilize a tonotopic (place) code, whereby FM is converted to AM through cochlear filtering. We measured FM and AM detection for a low f_c at slow (f_m = 1 Hz) and fast (f_m = 20 Hz) rates across three large groups: Young, normal-hearing (NH) listeners (n = 100), NH listeners varying in age (n = 85), and listeners varying in degree of sensorineural hearing loss (SNHL; n = 49). Results from all groups show high multicollinearity amongst FM and AM tasks. Data from SNHL but not NH listeners show strong correlations between the fidelity of cochlear place coding (frequency selectivity) and FM detection at both rates. Overall, the evidence suggests a unitary place code for FM. [Work supported by NIH Grant No. R01 DC005216.]

1:20

4pPP2. Tinnitus related to physiological changes in the auditory nerve in chronically noise-exposed mice. Laurel A. Screven, Kali Burke (Psych., Univ. at Buffalo, B80 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu), Amanda Lauer (Otolaryngol. HNS, Johns Hopkins Univ. School of Medicine, Baltimore, MD), Matthew A. Xu-Friedman (Biological Sci., Univ. at Buffalo, Buffalo, NY), and Micheal L. Dent (Psych., Univ. at Buffalo, Buffalo, NY)

Tinnitus is a pervasive auditory dysfunction affecting up to 10% of the adult population. The perception of ringing or hissing in the absence of a physical stimulus in one or both ears can be caused by acoustic trauma and other factors. Mice are a commonly used model for auditory disorders in humans, although the behavioral examination of tinnitus in mice has primarily been limited to reflexive measures involving inhibition of the acoustic startle by gaps in noise. Using an identification paradigm, we behaviorally tested whether mice show symptoms of tinnitus following long-term moderate noise exposure. Tinnitus was demonstrated by a shift in categorizing silence as narrowband noise. This experiment demonstrated that tinnitus can be induced in mice using noise exposure, similar to that caused by salicylate. Physiological and anatomical experiments reveal synaptic changes including response facilitation, increased reliability, increased synaptic terminal area, and increased number of release sites in mice housed in chronic background noise compared to standard-housed controls. These mice show normal auditory brainstem responses and a normal complement of peripheral auditory nerve synapses. This combination of behavioral and physiological methodologies enables researchers to examine both the behavioral manifestations and the underlying physiological mechanisms of tinnitus.

Contributed Papers

1:40

4pPP3. Neuronal frequency selectivity in the inferior colliculus and cochlear nucleus of the awake behaving macaque monkey. Jane A. Burton (Neurosci., Vanderbilt Univ., 111 21st Ave. S, 301 Wilson Hall, Nashville, TN 37240, jane.ann.burton@gmail.com) and Ramnarayan Ramachandran (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN)

Frequency selectivity relates to the ability to process complex signals and can be measured through auditory filters. Behavioral filters show broader tuning compared to cochlear and auditory nerve fiber tuning. To test whether filters evolve across the auditory pathway or if they are established in the periphery, we estimated neural filters in the cochlear nucleus (CN) and inferior colliculus (IC) and compared with simultaneously measured behavioral filters in macaques. Three macaques were trained to detect tones (signal = unit characteristic frequency (CF)) in spectrally notched maskers of varying width while single unit responses were recorded in the CN and IC. Filter shapes and bandwidths were estimated from the masked thresholds using the rounded exponential fit. Behavioral and neural filters increased in bandwidth with increasing CF. Behavioral and neural bandwidths were significantly correlated and not significantly different from each other for the CN and IC. Neural filter bandwidths were variable across units and structures, possibly reflecting heterogeneity of neuronal encoding strategies. These findings support a model in which behavioral frequency selectivity is established early in the auditory pathway. These data form the baseline for ongoing studies of macaques with noise-induced hearing loss and future studies of emerging hearing loss therapeutics.

1:55

4pPP4. Relations between speech perception in noise, high-frequency audiometry, and physiological measures of cochlear synaptopathy. Hannah Guest, Kevin Munro, and Christopher J. Plack (Manchester Ctr. for Audiol. and Deafness, Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester, Greater Manchester M13 9PL, United Kingdom, hannah.guest@manchester.ac.uk)

Cochlear synaptopathy, a loss of synapses between inner hair cells and auditory nerve fibers, is associated with age and noise exposure in animal models. However, the functional consequences of synaptopathy for humans are unclear. We pooled data from two recent studies to answer the question: are the common physiological measures of cochlear synaptopathy related to speech-perception-in-noise (SPiN) performance? Eighty-three audiometrically normal participants (ages 18–39) took part. Measures of synaptopathy were as follows: auditory brainstem response (ABR) wave I amplitude (102 dB peSPL click); ABR wave I:V amplitude ratio; envelope following response (EFR) amplitude (4 kHz carrier, 105 Hz modulation frequency); EFR amplitude growth with stimulus modulation depth; and middle ear muscle reflex threshold (1–4 kHz elicitors). We also conducted extended high-frequency (EHF) audiometry (10 and 14 kHz), suggested as a marker for synaptopathy in lower frequency regions. SPiN performance was assessed using the coordinate response measure with spatial maskers. None of the physiological measures of synaptopathy correlated significantly with SPiN. There was a significant correlation between EHF thresholds and SPiN, although it is unclear whether this is due to a direct relation between EHF hearing and SPiN, or whether elevated EHF thresholds are a marker for hidden damage at lower frequencies.

2:10

4pPP5. Olivocochlear system plasticity in response to chronic noise. Dillan F. Villavisanis, Katrina M. Schrode, Hamad A. Javaid (Dept. of Otolaryngol. - Head & Neck Surgery, Ctr. for Hearing and Balance, Johns Hopkins Univ. School of Medicine, 720 Rutland Ave., Baltimore, MD 21218, dvillav1@jhu.edu), Michael Muniak (Div. of Neurosci., Garvan Inst. of Medical Res., Sydney, NSW, Australia), Omobolade Odedoyin (Dept. of Otolaryngol. - Head & Neck Surgery, Ctr. for Hearing and Balance, Johns Hopkins Univ. School of Medicine, Baltimore, MD), Victoria Gellatly, Matthew A. Xu-Friedman (Dept. of Biological Sci., Univ. at Buffalo, Buffalo, NY), and Amanda Lauer (Dept. of Otolaryngol. - Head & Neck Surgery, Ctr. for Hearing and Balance, Johns Hopkins Univ. School of Medicine, Baltimore, MD)

Plasticity in the medial and lateral olivocochlear (MOC and LOC) systems in response to chronic exposure to background noise was investigated. Three different age-groups of young mice were exposed to the same chronic moderate noise conditions with either immediate tissue harvest or harvest following restoration to quiet conditions, with age matched controls. Cochleae were dissected and standard immunohistochemistry protocols were used to label hair cells with antibodies against myosin 6 and olivocochlear synaptic terminals with synaptic vesicle protein 2 (SV2). Specimens were imaged using confocal microscopy, and density of SV2 labeling was quantified. There was no statistically significant difference in MOC SV2 density between mice raised in noise and age matched controls for any group. However, there was a statistically significant increase in LOC SV2 density for adult mice raised in noise, particularly at higher frequency regions. This could suggest a protective upregulation of the efferent system against chronic moderate noise exposure. The increase in LOC innervation persisted for juvenile mice raised in noise with subsequent restoration to quiet conditions. These data suggest that the LOC system demonstrates sound-dependent plasticity, but that synaptic morphology may be altered for a substantial time period after exposure to noise ceases.

2:25

4pPP6. Reliability and interrelations of seven proxy measures of cochlear synaptopathy. Christopher J. Plack (Manchester Ctr. for Audiol. and Deafness, Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk), Hannah Guest, and Kevin Munro (Manchester Ctr. for Audiol. and Deafness, Univ. of Manchester, Manchester, Greater Manchester, United Kingdom)

Investigations of cochlear synaptopathy in living humans rely on proxy measures of auditory nerve function. Numerous procedures have been developed, typically based on the auditory brainstem response (ABR), envelope-following response (EFR), or middle-ear muscle reflex (MEMR). Some metrics correlate with synaptic survival in animal models, but translation between species is not straightforward; measurements in humans likely reflect greater error and greater variability from non-synaptopathic sources. The present study assessed the reliability of seven measures, as well as testing for correlations between them. Thirty-one normally hearing young women underwent repeated measurements of ABR wave I amplitude, ABR wave I growth with level, ABR wave V latency shift in noise, EFR amplitude, EFR growth with stimulus modulation depth, MEMR threshold, and an MEMR difference measure. Intraclass correlation coefficients indicated good-to-excellent reliability for the raw ABR and EFR amplitudes, and for both MEMR measures. The ABR and EFR difference measures exhibited poor-to-moderate reliability. No significant correlations, nor any consistent trends, were observed between measures, providing no indication that the between-subject variability in responses are due to the same underlying physiological processes. Findings suggest that proxy measures of cochlear synaptopathy should be regarded with caution, at least when employed in young, normally hearing adults.

2:40

4pPP7. Behavioral and physiological measures of auditory processing in individuals with autism spectrum disorder. Bonnie K. Lau (Univ. of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, blau@uw.edu), Ross K. Maddox (Univ. of Rochester, Rochester, NY), and Adrian K. C. Lee (Univ. of Washington, Seattle, WA)

Sensory processing abnormalities are a hallmark of autism spectrum disorder (ASD). In the auditory domain, hyper- and hypo-sensitivity to sound, reduced orientation to speech, and difficulty listening in noise are commonly reported. However, the etiology of these auditory processing abnormalities is poorly understood. In this preliminary study, multi-talker speech perception thresholds, otoacoustic emissions, electrophysiological responses, as well as standardized measures of cognition, language, and adaptive function were measured in each individual subject with ASD and their age-matched controls. Speech perception was assessed by estimating target-to-masker ratios at 50% correct for speech targets (0° azimuth) presented with two spatially separated (±45° azimuth) simultaneous speech maskers. The electrophysiological battery used to characterize the transmission and representation of sound included: (1) a click-evoked suprathreshold auditory brainstem response, (2) an envelope following response recorded to a 400-ms-long 4 kHz pure tone carrier amplitude modulated at 100 Hz at two modulations depths (0 and -6 dB) and (3) a binaural potential evoked by an interaural phase difference embedded in an amplitude-modulated carrier tone. Together, these measures provide a behavioral and physiological assay of auditory function and show the potential to further our understanding of the mechanisms underlying auditory abnormalities in individuals with ASD.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 SALON A (VCC), 1:15 P.M. TO 2:35 P.M.

Session 4pSCa

Speech Communication: Phonetics of Under-Documented Languages I

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Richard Wright, Cochair
Linguistics, University of Washington, Box 352425, Seattle, WA 98195-2425

Chair's Introduction—1:15

Invited Papers

1:20

4pSCa1. Phonetic documentation in the literature: Coverage rates for topics and languages. D. H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu), Christian DiCanio (Univ. at Buffalo, Buffalo, NY), and Rikker Dockum (Yale Univ., New Haven, CT)

With the introduction of automatic methods for examining both the acoustic and articulatory aspects of speech, our knowledge of the phonetic details of well-studied languages has grown substantially. Research on the phonetics of under-documented languages has expanded, but the breadth of coverage remains a challenge. What topics within the sound systems of these languages should be documented? With an ultimate aim of a reasonably complete typological assessment, should researchers focus on particular language families or phenomena and exclude others? Here, a survey of several hundred articles was undertaken, examining three major venues in the phonetics literature: *Journal of Phonetics*, *Journal of the International Phonetic Association*, and publications from the UCLA Languages of the World project (which appeared in various venues). 20 features were found to be addressed in many studies, and a rough guide to the coverage for a language can be seen in the total number of features reported in a study. Coverage has increased over time, but most studies still address a small percentage of the phonetic substance of the target language. Coverage across language family and region also increased but is still skewed toward Indo-European languages. Future directions and recommendations will be presented.

1:35

4pSCa2. Intervocalic consonant voicing vs. lengthening in two Athabaskan languages. Sharon L. Hargus (Dept. of Linguist, Univ. of Washington Seattle, Box 352425, Seattle, WA 98195-2425, sharon@uw.edu)

For over a century, various authors have described intervocalic consonants in Athabaskan languages as being long (e.g., Sapir 1914 on Chasta Costa, Cook 2004 on Dene Sųłiné (Chipewyan), McDonough and Ladefoged 1993 on Navajo). Such descriptions, even if instrumental, have generally not been very detailed, and have not adequately investigated the factors which may influence lengthening, such as stress and position in word. In this presentation, I first present results of an investigation of effects of position and stress on the duration of consonants with internal cues to duration (fricatives and consonantal sonorants) in two Athabaskan languages, Deg Xinag and Kwadacha Tsek'ene. Results show that intervocalic consonants are longer before stressed than unstressed syllables. Next, I present results of an instrumental study of [t] which shows that a competing and antagonistic process to intervocalic lengthening, intervocalic

voicing, is also found in both languages. Intervocalic stops are voiced through approximately 80% of their closure duration before unstressed vowels in both languages. Intervocalic voicing of stops has been reported for Tahltan (Bob 1997) and Dene Sų̨łiné (McDonough and Wood 2008) but generally not for other Athabaskan languages except for those in which nasals have evolved into voiced obstruents (e.g., Jicarilla Apache, Tuttle 2000).

1:50

4pSCa3. Kuki-Chin languages in Indiana: Investigating typologically rare sounds in a developing community of collaboration.
Kelly Berkson, Samson A. Lotven, James Wamsley, Zai Sung, Peng H. Thang, Thomas Thawngza, Kenneth de Jong, Sandra Kuebler, and Steven M. Lulich (Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

More than 20,000 Burmese refugees live in Indiana at the time of this writing. Most are originally from Chin State in western Myanmar and speak under-documented Tibeto-Burman languages from the Kuki-Chin branch. For the most widely spoken of them—Hakha Chin, also known as Laiholh or Hakha Lai—syntactic and morphological work exists but basic acoustic work is minimal. Others are completely undocumented: Zophei is spoken by about 4,000 people in Indiana and has to the best of our knowledge been mentioned but not described. In this talk, we describe a burgeoning collaboration between speech scientists and speakers of Kuki-Chin languages in Central Indiana. Our long-term communal goal is to develop resources ranging from the linguistic (e.g., documentary, analytical) to the practical (e.g., speech technology for use in emergency rooms, literacy materials). Phonetic investigation is a necessary precursor to much of this work, in part because these languages contain typologically unusual sounds (voiceless laterals, rhotics, and nasals; a 5-way coronal stop contrast involving dental and alveolar obstruents; voiced and voiceless laterally-released obstruents). Herein, we present the results of several phonetic investigations which will support future development of both theoretical and practical technological resources.

2:05

4pSCa4. A cross-linguistic study of phonetic correlates of metrical structure in under-documented languages.
Matthew K. Gordon, Ayla Applebaum (Dept. of Linguist, UC Santa Barbara, Santa Barbara, CA 93106, mgordon@linguistics.ucsb.edu), Thiago Chacon (Univ. of Brasilia, Brasilia, Brazil), Jack Martin (College of William & Mary, Williamsburg, VA), and Françoise Rose (Laboratoire Dynamique Du Langage, Université Lyon2/CNRS, Lyon, France)

Although languages with rhythmic stress have been extensively described and analyzed in the phonology literature, there has been little phonetic verification of the metrical structures suggested by these stress patterns. This paper presents results of a cross-linguistic study of the acoustic correlates of foot structure in several under-documented languages with morphologically complex words providing the necessary backdrop for the realization of rhythmic metrical structure. Various potential acoustic exponents of metrical structure are considered, including duration, F0, intensity, and formant frequencies. The targeted languages include Muskogean languages of the United States (Koasati, Muskogee, and Chickasaw), Circassian languages spoken primarily in Turkey and Russia (Kabardian and Adyghe), the Tukanoan language Kubeo of Brazil and Colombia, and the Arawak language Mojeño Trinitario of Bolivia. The languages differ in their foot templates (trochaic vs. iambic), the direction of their metrical parse (left-to-right vs. right-to-left), their degree of metrical rhythm, and whether they also possess lexical tone or not. Results suggest considerable diversity in the acoustic manifestations of metrical structure (including the possibility of lack of rhythmic feet even in long words) and in the relationship between the word-level metrical system and other prosodic features, including intonation and (in tonal languages) lexical tone.

2:20

4pSCa5. Obstruent and rhotic contrasts in Adnyamathanha, a language of South Australia.
Andrew Butcher (Speech Pathol. & Audiol., Flinders Univ. of South Australia, Flinders University, GPO Box 2100, Adelaide, SA 5001, Australia, andy.butcher@flinders.edu.au) and John McEntee (Independent Researcher, Glenelg South, SA, Australia)

Adnyamathanha is one of the Thura-Yura languages, spoken in the northern Flinders Ranges of South Australia. It has a fairly complex consonant system with six places of articulation (including four coronals). Through a combination of traditional phonological analysis and acoustic phonetic measurement, we attempt to throw some light on one aspect of this complexity. We show that there is a contrast between two series of obstruents in intervocalic position, which sets Adnyamathanha apart from most other languages of the region—and of the Pama-Nyungan language family as a whole. The main phonetic correlates of the contrast are closure duration and presence versus absence of glottal pulsing during the closure. Voiceless obstruents are consistently realised as long stops, whereas their voiced counterparts, though always shorter, vary in duration and manner of articulation, depending on place of articulation. We analyse the voiced labial fricative as the voiced equivalent of the voiceless (bi)labial stop and we analyse the alveolar tap and the retroflex flap as voiced cognates of the voiceless alveolar and retroflex stops respectively. Thus, although we recognise the existence of four phonetically distinct rhotic sounds, we assign only the alveolar trill and the retroflex glide to the phonological category of rhotics.

Session 4pSCb

Speech Communication: Phonetics of Under-Documented Languages II (Poster Session)

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

Richard Wright, Cochair
Linguistics, University of Washington, Box 352425, Seattle, WA 98195-2425

All posters will be on display from 3:00 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 3:00 p.m. to 4:00 p.m. and authors of even-numbered papers will be at their posters from 4:00 p.m. to 5:00 p.m.

Contributed Papers

4pSCb1. The rhythm of Hul'q'umi'num'. Mackenzie Marshall and Sonya Bird (Dept. of Linguist, Univ. of Victoria, 3800 Finnerty Rd, Victoria, BC V8P 5C2, Canada, mackenzieomarshall@gmail.com)

This research explored the linguistic rhythm of Hul'q'umi'num', a dialect of the Coast Salish Hul'q'umi'num'-Halq'eméylem-hən̓q̓əmin̓əm̓ language, and initiated phonetic documentation of it. Rhythm was analyzed from an audio file of an Elder telling a story. The story was segmented and phonetically transcribed using acoustic analysis software (Praat), and rhythm was measured based on the segmentation. Rhythm metrics demonstrated that consonantal intervals of Hul'q'umi'num' patterned like no other documented language (according to ΔC and VarcoC). In terms of vocalic intervals (%V, ΔV, and VarcoV), Hul'q'umi'num' patterned in the same rhythmic category as English ("stress-timed"). Interestingly, several differences emerged between the segmentation-based phonetic transcription and the transcription provided by language experts, such as consonant cluster elisions, loss of glottal stops, and vowel alternations. Further investigation of their systematicity and effects on Hul'q'umi'num' rhythm as a whole is needed to understand what components of the language give it its unique rhythm.

4pSCb2. Pitch realization of post-focus components in Chongming Chinese. Yike Yang, Si Chen (Dept. of Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., AG511, Kowloon N/A, Hong Kong, yi-ke.yang@connect.polyu.hk), and Kechun Li (The Univ. of Hong Kong, Hong Kong, Hong Kong)

Prosodic focus makes use of pitch to highlight part of an utterance. It has been generally found that a focused component is realized with an expanded pitch range, and there is evidence suggesting that a post-focus component may be associated with a reduced or compressed pitch range, the phenomenon of which is further coined as post-focus compression (PFC). However, the presence or absence of PFC seems to diverge even within the same language family, and what is more interesting is the potential interaction between focus and lexical tones in tone languages. The current project thus aims to investigate whether PFC is present and how post-focus pitch is realized in Chongming Chinese, an under-documented language with eight tones. Specifically, we conducted a production experiment in which 20 target words varying in consonants, vowels and tones were selected, four focus conditions (no focus, pre-focus, on focus and post-focus) were designed, and four tonal contexts with different proceeding and following syllables were manipulated. Linear mixed-effects models were fitted to examine the effects of these variables on the realization of pitch contours in the post-focus components and also tackle the interactions among them. Our findings contribute novel data to the prosodic typology literature.

4pSCb3. Dagaare [a] is not neutral to ATR harmony. Avery Ozburn, Samuel Akinbo, Alexander Angsongna, Murray Schellenberg, and Douglas Pulleyblank (Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC, Vancouver, BC V6T 1Z4, Canada, aozburn@gmail.com)

Bodomo (1997) describes Dagaare (Gur; Ghana) as having a single low vowel, [a], which is neutral to ATR harmony. This paper presents acoustic data from a study of Dagaare <a> which is inconsistent with this description. A list of sentences was elicited from five native speakers of Dagaare. Each sentence contained <a> in one of four verbal particles situated in one of four contexts: ATR _ ATR, ATR _ RTR, RTR _ ATR, and RTR _ RTR. Formants of the low vowel were measured and compared across contexts. Results showed a substantial, significant difference in F1 values and a smaller but still significant difference in F2 values in contexts where <a> is followed by an ATR word compared to when it is followed by an RTR word. All speakers and all particles showed the same pattern. We conclude that, contrary to previous claims, the Dagaare low vowel is not neutral to harmony, but rather has acoustically distinct variants in RTR versus ATR contexts. Bodomo, A. (1997). *The structure of Dagaare.* California: CSLI publications. [Funded by SSHRC.]

4pSCb4. Phonetic structure of Intonational prominence in 3 Dene/Athabasakn (ISO den) languages. Joyce McDonough (Dept. of Linguist, Univ. of Rochester, Rochester, NY 14627, joyce.mcdonough@rochester.edu)

In this paper, we examine the prosodic structure of Y/N Q's and focus constructions in three related Dene/Athabaskan tone languages spoken in North American: Navajo, Dene Su̜łı́ne, and North Slavey (Deline), known for their complex polysynthesis, representing a geographically widespread language family, from the American Southwest (Navajo), to northern Alberta (Dene Su̜łine) to the arctic (North Slavey). These languages share a strikingly similar phonetic inventory and morphological structure. One interesting feature is the phonotactic structure of the morphology—the monosyllabic verb stem is the location of phonemic contrasts: outside the stem the contrasts are severely reduced, making the stems phonemically and phonetically prominent. The stems are the rightmost elements in the verbal complex and in a sentence (verb final). Navajo has been argued to lack any kind of intonational prominence, but the northern languages show more typical prominence marking. In this study we lay out the difference associated to the demarcation of boundaries and prominence: duration and the acoustic correlates of pitch accent events associated with focus and YN/Q's and boundary marking. Data is taken from conversation games models on map tasks. Findings indicate while phonetically and morphologically similar great variety is found in the prosody.

4pSCb5. Neutralization of underlying vs. derived "unnatural" palatals in Xhosa. Aaron Braver (Texas Tech Univ., P.O. Box 43091, Lubbock, TX 79409-3091, aaron@aaronbraver.com)

Xhosa (Bantu) has an "unnatural" process of palatalization—contrary to typological expectations, it is triggered by [w], but not [i] or [j], and applies only to labials. For example, [m]→[ɲ] as in uku-lum-à to bite' ~ uku-luɲ-w-a 'to be bitten', and [mb]→[ⁿd͡ʒ] as in uku-ɬamb-a 'to wash' ~ uku-ɬaⁿd͡ʒ-w-a 'to be washed'. This paper compares underlying and derived palatals to determine whether the palatal/non-palatal contrast is neutralized (in)completely in the derivational context. 40 Xhosa nonce verbs, ending in segments which undergo palatalization, were created to be read by 6 native speakers. Speakers read these forms and were then asked to produce forms with the passive -w suffix, thus triggering palatalization. F2 was measured at boundaries and ten ms. into the vowels preceding and following the palatalized segment. F2 at ten ms into the following vowel differed on average by 167.99 Hz between underlying and derived palatals, trending towards, but not reaching significance ($t = 1.94$, $p = 0.054$), while F2 slope showed no clear trend. While Zsiga (1995) proposes that English palatalization can be complete or gradient, this is the first study of palatalization in the context of incomplete neutralization, and the first study to look at incomplete neutralization in Bantu.

4pSCb6. The special nature of Australian phonologies: Why auditory constraints on the sound systems of human languages are not universal. Andrew Butcher (Speech Pathol. & Audiol., Flinders Univ., GPO Box 2100, Adelaide, SA 5001, Australia, andy.butcher@flinders.edu.au)

The phonological systems of human languages are constrained by what are often assumed to be universal properties of human auditory perception. However, the atypical phonologies found in many hearing-impaired speakers indicate that such constraints also operate at an individual level. The phonology of a child with chronic *otitis media*, for example, may lack a voicing distinction or sometimes have no fricatives. So, if a large group of speakers in a speech community operates with an atypical auditory system over many generations, then the phonology of the language(s) spoken by such a community might also over time be influenced by the particular properties of that common auditory system. Over half of the Australian Aboriginal population develop chronic *otitis media* with effusion in infancy and 50–70% of Aboriginal children have a significant hearing loss at both ends of the frequency range. Most Australian languages have phonologies which are atypical in world terms, having no voicing distinction and no fricatives or affricates, but an unusually large number of places of articulation. Acoustically, the sound systems of Australian languages appear to be a very good match for the hearing profiles of large numbers of their speakers. This paper reviews the evidence for a connection.

4pSCb7. Acoustic properties of singleton and geminate ejective stops in Tsova-Tush. Bryn G. Hauk (Linguist, Univ. of Hawaii at Manoa, 713 Hausten St. #2, Honolulu, HI 96826, bhauk@hawaii.edu)

Tsova-Tush, also known as Batsbi, is an underdescribed Northeast Caucasian language spoken by a few hundred people in Zemo Alvani, Georgia. The Tsova-Tush phoneme inventory includes four geminate stops that contrast with singletons at the same place of articulation: /tʰː t'ː qʰː q'ː/. The existence of geminate ejective stops is particularly interesting, as such phonemes are cross-linguistically rare, having been reported in only 13 of the 2,155 phoneme inventories sampled by PHOIBLE (http://phoible.org/). The present study characterizes these stops in Tsova-Tush based on high-quality audio recordings of four Tsova-Tush speakers producing a list of 65 target words in a carrier sentence. The following measures were statistically compared: closure duration, VOT, duration of the preceding vowel, and f_0 and H1*-H2* in the following vowel. Preliminary results from three speakers suggest that ejectives are characterized by shorter VOT, a shorter preceding vowel, and a difference in both f_0 and H1*-H2* in the following vowel, with marked interspeaker variation in the latter measures. This study provides the first detailed description of this typologically interesting two-by-two contrast (singleton aspirates, geminate aspirates, singleton ejectives, geminate ejectives) at two places of articulation (dental and uvular) in an underdescribed language.

4pSCb8. An acoustic comparison of /θ/ and /s/ in Comox-Sliammon. Gloria Mellesmoen (Univ. of Br. Columbia, Box #346 - 6335 Thunderbird Crescent, Vancouver, BC V6T 2G9, Canada, gloria.mellesmoen@alumni.ubc.ca)

Though some Coast Salish languages have innovated /θ/, a typologically rare segment, the only study of Salish fricatives describes Montana Salish, an Interior language without /θ/ [Gordon *et al.* (2002). A cross-linguistic acoustic study of voiceless fricatives. *JIPA, 32*(2), 141-174.]. In addition to being typologically noteworthy, /θ/ is also interesting within Salish, as impressionistic descriptions suggest articulatory similarity and perceptual ambiguity between /θ/ and /s/. This is found in Halkomelem, Northern Straits, and Comox-Sliammon. Motivated by a gap in documentation and reported ambiguity, this paper is an acoustic study of /θ/ and /s/ in Comox-Sliammon. Following the methodology of Reidy [(2016). Spectral dynamics of sibilant fricatives are contrastive and language specific. *JASA, 140*(4), 2518–2529.], PeakERB$_N$ trajectories are compared for /θ/ and /s/ across four fluent speakers of Comox-Sliammon. The results suggest that the fricatives are acoustically distinct, though there is considerable inter-speaker and intra-speaker variability for /θ/. Lack of overlap for three of four speakers suggests that the source of the reported ambiguity may be L1 English speaker perception, rather than the realization of Mainland Comox fricatives. The high level of variation suggests that /θ/ may be a recent and unstable innovation, supporting the reconstruction of a Proto-Comox [s]-like form.

4pSCb9. Pausing as a prosodic correlate of speech units in St'át'imcets (Lillooet Salish). Marion Caldecott, Ewa Czaykowska-Higgins, Janet Leonard, and John Lyon (Office of Linguist, Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, mariongc@uvic.ca)

The reliability of pausing as a correlate to prosodic, syntactic and discourse units has been debated in commonly-studied languages (outlined, for instance, in Krivokapic 2007). Preliminary research in Nxaʔamxčín (Interior Salish) (Caldecott & Czaykowska-Higgins 2012) has shown that pausing could be a more reliable acoustic correlate of these structures than pitch, in line with previous research indicating that Salish languages do not exhibit a strong reliance on pitch to mark information structure (Caldecott 2017; Caldecott & Czaykowska-Higgins 2012; Davis 2012; Koch 2008, 2011). The current study further tests this hypothesis by examining the occurrence and duration of pauses in a 9.5-minute spontaneous narrative by a fluent speaker of St'át'imcets (Interior Salish). Pause distribution with respect to syntactic and prosodic boundaries is analysed. Preliminary analysis suggests that St'át'imcets follows previous research on spontaneous narratives in some respects but not others. Very long pauses (<5.5 s) mark major thematic shifts (Oliveira 2000) and most pauses (70%) occur clause-finally (Henderson, Goldman-Eisler & Skarbek 1966). Most clause-medial pauses occurred between a determiner and noun, following Goldman-Eisler (1968), but contra Gee & Grosjean (1983). A t-test indicates that clause-final vs. clause medial pauses are not significantly different in duration ($p > .05$) (contra Goldman-Eisler 1972).

4pSCb10. An acoustic and articulatory investigation of the Mauritian vowel inventory. Samantha Myers (Lingustics, Indiana Univ., Bloomington, IN), Fabiola Henri (Univ. of Kentucky, Lexington, KY), and Kelly Berkson (Lingustics, Indiana Univ., 1021 E. Third St., Dept. of Linguist - Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

While there have been a decent amount of morphosyntactic and sociolinguistic studies on Mauritian, a "French-based" creole spoken by most of the population of Mauritius (e.g. Baker 1972, Alleesaib 2012, Henri 2010, Miller 2015, Syea 2013), phonetic or phonological description on the language hardly exists. Pudaruth (1993) proposes a phoneme inventory consisting of 19 consonants and 8 vowels, intuitively noting that rhotics preceding vowels are weakened, almost unpronounced. In the present work, we empirically investigate the vowel inventory of Mauritian using acoustic and articulatory data. In addition to presenting basic acoustic measures (e.g. vowel spaces), we investigate the rhotic and report acoustic measures which suggest strong variation between the orthographic "a" that precedes "r" and that which precedes all other written consonants. Ultrasound imaging confirms that the pre-rhotic "a" differs from other "a"s articulatorily.

4pSCb11. Acoustics of Tatar vowels: Articulation and vowel-to-vowel coarticulation. Jenna Conklin and Olga Dmitrieva (Linguist, Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu)

Volga Tatar is a Turkic language spoken by 5 million people in Central Russia for which instrumental acoustic descriptions are lacking. This study uses formant analysis of acoustic recordings from 27 native speakers of Volga Tatar to describe the vowels of Tatar and evaluate the accuracy of previous impressionistic phonetic descriptions. In addition to describing the acoustic vowel space of Tatar, the study uses carefully chosen target words to evaluate vowel-to-vowel coarticulation in height and backness among the Tatar vowels /i, æ, ɑ/. Examining vowel-to-vowel coarticulation in Tatar is of particular theoretical interest due to the presence of vowel harmony in the language. While the majority of native Tatar words are subject to backness, and possibly rounding, harmony, a large class of disharmonic lexical items, mostly from borrowings, provides insight into the coexistence of long-distance phonological vowel assimilation (vowel harmony) and long-distance phonetic vowel assimilation (coarticulation) in the same language. Previous research (Banerjee, Dutta, & S., 2017; Beddor & Yavuz, 1995) suggests that vowel harmony may suppress coarticulation proceeding in the same direction. However, the current results indicate that direction of coarticulation in Tatar is mediated primarily by other factors, such as target and trigger vowel identity.

4pSCb12. Variation in Scottish Gaelic preaspirated stops: Across speech styles and in connected speech. Maya Klein (Linguist and Anthropology, Univ. of Arizona, 1103 e University Blvd., Tucson, AZ 85721, mayaklein@email.arizona.edu)

This paper examines variation in preaspiration in Scottish Gaelic, an endangered language spoken in Scotland. Previous studies (Clayton 2011; Nance & Stuart-Smith 2013) have examined Gaelic preaspiration in careful speech. The present study investigates whether 1. Preaspiration patterns similarly in casual speech as previously attested in careful speech and 2. If variation in preaspiration resembles processes of stop reduction, or sound change in progress. Preaspiration is examined in casual speech across speakers, and across different speech styles within one speaker. Preaspiration is measured for duration, as well as band-pass filtered zero crossing rates, developed by Gordeeva & Scobbie (2010) and adapted for Scottish Gaelic by Nance & Stuart-Smith (2013). Results confirm that variation in preaspiration pattern similarly to previous studies: preaspiration is longer and has higher change in zero crossings for /k/ and in words with a preceding short vowel. This study contributes to the existing literature in that it shows that preaspiration is shortest in words that are phrase medially. This study shows that preaspirated stops are not heavily reduced, and that variation in preaspiration patterns like an ongoing sound change. This study highlights the importance of carefully investigating variation in typologically rare phenomena in endangered languages.

4pSCb13. Perception of focus in Sümi (Tibeto-Burman, India). Amos Teo and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, 1505 Orchard St., Unit 28, Eugene, OR 97403, ateo@uoregon.edu)

This paper investigates the homophony/polysemy between a morphological agentive marker and a contrastive focus marker in Sümi, a Tibeto-Burman language of Northeast India. Both are realized by an enclitic =*no* that attaches to grammatical subjects, but the interpretation of the enclitic varies by clause type: textual analysis shows that =*no* with subjects of transitive clauses is more commonly interpreted as an agentive marker without focus; while =*no* with subjects of verbless and intransitive clauses is more commonly interpreted as a contrastive focus marker (Teo 2018). The present study examines whether transitive subjects in contrastive focus receive any special prosodic marking that is recognizable to native listeners. We answer this question using a ratings task in which listeners were presented with sentences that were produced under different pragmatic conditions to have broad focus over the entire clause or narrow contrastive focus on the subject, i.e., in responses to a question like "Who (out of a set of potential participants) did the action?" The study has implications for understanding the development of focus markers in other languages of the Himalayas, as well as in New Guinea and Australia where homophony/polysemy between agentive or ergative markers and focus markers has been found.

4pSCb14. A vowel space comparison of Tlawngrang Zophei and Lawngtlang Zophei. Samson A. Lotven and Kelly Berkson (Dept. of Linguist, Indiana Univ., 1020 E. Kirkwood Ave., Bloomington, IN 47405-7005, slotven@indiana.edu)

Zophei refers to an undescribed group of Tibeto-Burman languages within the Kuki-Chin family. Originally spoken in the Chin Hills of Western Myanmar, approximately 4,000 Zophei-speaking refugees now live in Central Indiana. No previous research on Zophei exists. The speakers located in Indiana who identify as ethnically Zophei hail from 14 distinct villages, and it is not yet known how many dialects or languages are represented. As part of a larger effort to kick-start a research program on Zophei, the current study compares and contrasts the vowel spaces of two speakers, one from Tlawngrang and one from Lawngtlang. The vowel spaces created for each speaker show some clear differences, especially with regards to the number and distribution of high vowels and diphthongs, indicating that these two areas speak different varieties with markedly different phonologies. For example, where one speaker has an /ui/ diphthong the other speaker r consistently has the front rounded monophthong /y/. This research contributes to our ultimate goal, which is to determining the dialectal make-up of Zophei and to develop a description of the language or languages spoken by the ethnic Zophei population in Indiana.

4pSCb15. F0 of the verbal nasal prefix in Medumba. Yadong Liu (Linguist, Univ. of Br. Columbia, Run Run Shaw Tower, The University of Hong Kong, Hong Kong 000000, Hong Kong, yadong@connect.hku.hk)

Both tone-bearing unit (TBU) and non-TBU syllables can exist in a tonal language [Roberts 2003, *Cahiers voltaïques/Gur Papers* 6, 95]. The verbal nasal prefix bears a high tone in many Bamileke languages [Voorhoeve 1974, *Stud. Afr. Ling.* 5: 205]; in Medumba, however, anecdotal evidence suggests that the nasal prefix may be a non-TBU [Keupdjio 2018, *PC*] in which case it cannot bear a high tone. The present study investigates the F0 contour of the verbal N-prefix and its surrounding TBUs (e.g., vowels in the preceding auxiliary verb and the following main verb) in various contexts in Medumba. Acoustic results show that the F0 contour of the nasal prefix itself is largely determined by the F0 trend of the preceding vowel, and no F0 change is observed at the onset of the nasal prefix, confirming that the verbal N-prefix is a non-TBU in Medumba. However, the nasal prefix appears to induce a raised F0 at the onset of the following TBU, indicating that, although the N-prefix itself does not express tone, it nevertheless "bears" a high tone. Implications for theories of TBUs will be discussed.

4pSCb16. Enriching the understanding of glottalic consonant production: Vertical larynx movement in Hausa ejectives and implosives. Miran Oh, Dani Byrd, Louis Goldstein (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu), and Shrikanth S. Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Ejectives and implosives are understood to have laryngeal activity distinct from their pulmonic counterparts. However, while correlations of vertical larynx movement (VLM) with tone and intraoral pressure are known, little articulatory data exists regarding VLM in glottalic consonants. This study examines the degree and timing of VLM in a Hausa speaker's productions, using real-time MRI of the vocal tract. We use a novel centroid tracking technique developed to quantify VLM. Findings indicate that the onset of VLM is tightly coordinated with oral closure achievement for glottalic stops. For ejectives, larynx raising is faster than in plosives and is especially large for ejective fricatives, which require maintenance of sufficient air flow for turbulence. Additionally, contrary to previous descriptions, implosives and voiced plosives do not differ in degree of larynx lowering. Instead, they contrast in timing of the VLM with the oral closure; larynx lowering starts early, with initiation of the oral constriction gesture, for voiced plosives; whereas lowering begins later, at oral closure achievement, for implosives. Overall these patterns suggest that VLM may be coupled in an anti-phase mode with the oral constriction gesture in both glottalic stops and coupled in-phase with the oral gesture in pulmonic stops. [Work supported by NIH.]

4pSCb17. The production and perception of Ikema geminates. Catherine Ford and Benjamin V. Tucker (Univ. of AB, Unit 204 7111 80 Ave. NW, Edmonton, AB T6B0C7, Canada, cford1@ualberta.ca)

Gemination is largely assumed to be cued through duration. While production and perception studies consistently confirm the importance of duration as a cue, certain phones are not as easily distinguished using duration alone. The current study analyzes geminate and singleton minimal pair production and perception in Ikema, an endangered Japonic language, to determine to what extent other cues may contribute to this distinction. Productions of stop, fricative, affricate, nasal, and rhotic geminates and singletons were used to determine acoustic differences and possible cues to gemination. We analyzed duration for all phones, and spectral moment and vocalic measures for fricatives and affricates. Results indicate that while duration is an important cue, spectral moment measures and vocalic measures are better predictors of affricate geminates and contribute significantly to the acoustic distinction among fricatives. Word-initial stops may be difficult to identify as closure duration is a more significant predictor of gemination than voice onset time. Listener perception was analyzed using a lexical decision task. Results indicate that while Ikema speakers distinguish geminates from singletons in novel situations, non-word perception may not be consistent cross-linguistically due to varying exposure to the concept, and community-internal acceptability of pronunciation variation.

4pSCb18. Comparative analysis of South Korean and North Korean vowels: A pilot study. Jungah Lee, Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, 1455 Moss St. Unit 306, Graduate Village Apartment, Eugene, OR 97403, jlee27@uoregon.edu), and Charlotte Vaughn (Linguist, Univ. of Oregon, Eugene, OR)

This study investigates cardinal vowels of standard North Korean and South Korean. Prior reports have suggested that North and South Korean vowels have undergone changes after decades of relative isolation. This poster reports a pilot study investigating the ways in which the language standards of North and South Korea are similar and different by examining the speech of newscasters from each country. Acoustic analysis of the speech data suggested that North Korean vowels [ɛ] and [æ] were produced in the higher position relative to the South Korean counterparts, and the back vowels [ʌ] and [o] showed overlapping formant values unlike the South Korean counterparts. The perception experiment suggested that South Korean listeners could not accurately identify the North Korean [ʌ] and [o]. These results indicate that there may be interesting differences across North Korean and South Korean vowels.

4pSCb19. On the non-universality of intonation: Evidence from Triqui. Christian DiCanio and Richard Hatcher (Linguist, Univ. at Buffalo, 601 Baldy Hall, Buffalo, NY 14260, dicanio@haskins.yale.edu)

Languages with large lexical tone inventories typically involve less freedom for suprasegmental properties to be manipulated for pragmatic meaning or phrasal constituency (Connell 2017). However, such languages may still use F_0 to a limited degree for marking information structure or utterance finality (DiCanio *et al.* 2018, Xu 1999). We present results from three field experiments with 11 speakers where we investigated information structure and prosody in Itunyoso Triqui, an Otomanguean language (Mexico). Itunyoso Triqui possesses nine lexical tones (/4, 3, 2, 1, 43, 32, 31, 13, 45/), fixed final stress, and contrastive phonation type. In experiment 1, we examined tone production in words in broad and narrow focus contexts. Words under narrow focus were lengthened slightly (13-14%) but no general effect of focus on F0 levels or contours was found. In experiment 2, we examined tones in utterance non-final and final contexts. Words were lengthened in utterance-final position relative to non-final position, but no F0 differences were found. In experiment 3, we investigated F0 declination in sentences consisting of only level tones and found no F0 change across utterances. The results from these experiments suggest that Itunyoso Triqui does not use F_0 to encode information structure or prosodic boundaries.

4pSCb20. Vowel length distinctions in Plains Cree. Angeliki Athanasopoulou and Darin Flynn (School of Lang., Linguist, Literatures, and Cultures, Univ. of Calgary, Craigie Hall D310, 2500 University Dr. N.W., Calgary, AB T2N1N4, Canada, angeliki@udel.edu)

Plains Cree is a widely-spoken Indigenous language in Canada. Its vowels are traditionally described as contrasting short /i, o, a/ vs. long /i:, o:, a: (e:)/ (Wolfart 1973). Muehlbauer's (2012) acoustic study of speakers born in the early 20th century confirms that duration is a significant cue, but differences in vowel quality were just as significant. We investigate the durational and quality differences in a younger L1 speaker's productions of short and long vowels. Preliminary analysis of 195 vowels shows that the durational difference between the long (102ms) and short vowels (66ms) remains significant on average, but there is greater overlap between the two length categories than previously thought: the range is 40-186ms for long vowels and 15-201ms for short ones. Vowel quality remains significant, too. In addition, we report on the results of a logistic regression analysis that tests the relative importance of duration vs. quality (and their interaction) as acoustic cues to the phonological contrast in the current generation of Plains Cree speakers. Our findings dovetail with descriptions of Eastern varieties, e.g.: "the old long/short distinction in Proto-Cree became (or is still becoming) a tense/lax distinction in East Cree" (Dyck 2011).

4pSCb21. Stop, approximant, and timing slot: The changing faces of the velar stop in Iwaidja. Jason Shaw (Yale Univ., Dow Hall 305, New Haven, CT, underlying.representation@gmail.com), Christopher Carignan (LMU Munich, Munich, Germany), Tonya Agostini (Univ. of Newcastle, Wollongong, NSW, Australia), Robert Mailhammer (Humanities and Commun. Arts / The MARCS Inst., Western Sydney Univ., Penrith, NSW, Australia), Mark Harvey (Univ. of Newcastle, Newcastle, NSW, Australia), and Donald Derrick (Univ. of Canterbury, Christchurch, New Zealand)

Limited access to speakers and incomplete lexical knowledge are common challenges facing phonetic description of under-documented languages. We address these challenges by taking a multi-dimensional approach, seeking to constrain our phonetic description by covariation across acoustic and articulatory parameters. We demonstrate the approach through an analysis of velar consonants in the Australian Aboriginal language Iwaidja. Existing accounts contrast a velar stop /k/ with a velar approximant /ɰ/ in word-medial position (Evans 2009). Converging evidence from ultrasound images of the tongue body and acoustic analysis of intensity data reveal that the posited opposition is not consistent across speakers (N = 4) and lexical items. Unsupervised categorization of the phonetic data indicates two phonetic categories, appropriately labelled as [a] and [ɰ], which do not map consistently to dictionary labels in existing descriptions. We conclude that speaker-specific allophonic variation is the result of an ongoing process of lenition of /k/ between sonorant segments which has not yet phonologized. More broadly, integrating phonetic dimensions revealed categories that were ill-defined on the basis of just acoustic or articulatory measures alone. Depth of analysis, characterized by phonetic multi-dimensionality, may support robust generalization where broad analysis (multiple speakers, large corpora) are impractical or impossible.

4pSCb22. Laryngeal contrasts in first and second language speakers of Hul'q'umi'num'. Maida Percival (Linguist, Univ. of Toronto, 100 St. George St., 4th Fl., Toronto, ON M5S 3G3, Canada, maida.percival@mail.utoronto.ca), Sonya Bird (Linguist, Univ. of Victoria, Victoria, BC, Canada), and Donna Gerdts (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

This paper investigates laryngeal contrasts in first (L1) and second (L2) language speakers of Hul'q'umi'num', a dialect of Halkomelem (Salish). The language, which lacks a thorough acoustic description of its consonants, is highly endangered with 50-100 native speakers, but has a growing number of L2 speakers. Since the learners come from an English background and are therefore unfamiliar with glottalized consonants, the sounds can prove challenging. This study addresses this issue by examining how L1 and L2 speakers pronounce plain and ejective stops in terms of what acoustic correlates they're using. Tokens of words read in isolation will be

analyzed from L1 and L2 speakers participating in a language course in Duncan, BC. Tokens of both speaker groups will be classified by phoneme and acoustic measures of duration (e.g., voice onset time, closure duration), centre of gravity, and vowel coarticulation (e.g., spectral tilt, F0, rise time) will be made. The results of the two groups will be compared with statistical analysis, and the findings will be used to create guidelines to assess future learners' and teach pronunciation. The findings will also contribute to a broader understanding of how Hul'q'umi'num' consonants fit into voicing and ejective typology.

4pSCb23. The Suzhou Chinese "apical" and "fricative" vowels: Uniformity and idiosyncrasy. Matthew Faytak (Linguist, Univ. of California, Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

"Fricative" and "apical" vowels have been variously described as syllabic voiced fricatives, syllabic approximants, or high vowels, often based on impression. To better characterize their articulatory and acoustic nature, ultrasound video and acoustic data on the production of the "fricative" and "apical" vowels were collected from 43 speakers of Suzhou Chinese, along with the vowels [i, y, æ, u] and the fricatives [s, ɕ]. Analysis reveals substantial interspeaker variation, largely idiosyncratic, which may contribute to the range of descriptions given to "fricative" and "apical" vowels. This variation, however, obscures the fact that the "fricative" vowels are curiously uniform in tongue posture with the alveolopalatal fricatives for most speakers. The "apical" vowels are consistently produced with a tongue posture similar to the alveolar fricatives and affricates, such as [s], that they obligatorily follow. The "fricative" vowels, on the other hand, are phonotactically freer, and speakers exhibit a wider range of articulatory strategies. Individuals' favored articulations can be situated on a continuum between lamino-postalveolar, or [ɕ]-like, and dorso-postalveolar, akin to an advanced [i]; most speakers favor the lamino-postalveolar strategy. Acoustic analysis suggests that speakers also vary in whether they favor production of fricative noise during both the "fricative" and "apical" vowels.

THURSDAY AFTERNOON, 8 NOVEMBER 2018 — RATTENBURY A/B (FE), 1:00 P.M. TO 4:25 P.M.

Session 4pSP

Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics: Detection and Tracking of Mobile Targets II

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

Kainam T. Wong, Cochair
Dept. of Electronic & Information Engineering, Hong Kong Polytechnic University, 11 Yuk Choi Road, Hung Hom KLN, Hong Kong

Contributed Papers

1:00

4pSP1. Post-tracking range-rate estimation with up/down linear-frequency-modulated pulses. Douglas Abraham (CausaSci LLC, PO Box 627, Ellicott City, MD 21041, abrahad@ieee.org)

Range-rate (RR) estimation in active sonar is most often accomplished using continuous-wave (CW) pulses. An alternative that is more suited for use with slowly moving objects in reverberation-limited conditions is presented and evaluated. As described by A. Rihaczek [Principles of High Resolution Radar, McGraw Hill, 1969], the approach entails simultaneously transmitting up- and down-sweeping linear frequency modulated pulses (LFM-U&D). The key enabler of the LFM-U&D is the complementary range-bias information relative to using a single LFM sweep type (LFM-U|D). Novel in the proposed LFM-U&D approach is the use of multiple pings after tracking to form improved range and RR estimates. Cramer-Rao lower bounds (CRLBs) on estimator variance as a function of the number of pings illustrate that LFM-U&D is a compromise between the CW and the standard LFM-U|D. Estimates of the RR of clutter tracks from the Target and Reverberation Experiment (TREX13) were used to evaluate the approach. An order of magnitude reduction in the inter-quartile range (IQR) of the RR estimate was observed for LFM-U&D relative to LFM-U|D, which allows a significant reduction in the number of false tracks. The full reduction in IQR predicted by the CRLB was seen to be limited by the effects of surface motion.

1:15

4pSP2. Acoustic tracking and localization using computationally efficient algorithms. Geoffrey H. Goldman (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

U.S. Army Research Laboratory (ARL) is developing low size, weight, power, and cost (SWAP-C) solutions for acoustic detection, tracking, and localization of moving targets. Many standard signal processing algorithms cannot be implemented on low SWAP-C commercial off-the-shelf (COTS) hardware for unattended ground sensor applications due to power and computational requirements. To overcome this issue, ARL is developing lower complexity beamforming, tracking and localization algorithms. For example, beamformer algorithms are based upon 1-dimensional scans, multi hypothesis target trackers are based on alpha beta filters and localization algorithms are based upon linear least squares algorithms. The performance of these algorithms is slight poorer than other standard approaches, but their computational complexity can be lower by an order of magnitude.

Invited Papers

1:30

4pSP3. DOA estimation in heteroscedastic noise. Peter Gerstoft, Kay L. Gemba (Noise Lab, Univ. of California, San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu), and Santosh Nannuru (Noise Lab, Univ. of California, San Diego, San Diego, CA)

The paper considers direction of arrival (DOA) estimation from long-term observations in a noisy environment. In such an environment the noise source might evolve, causing the stationary models to fail. Therefore a heteroscedastic Gaussian noise model is introduced where the variance can vary across observations and sensors. The source amplitudes are assumed independent zero-mean complex Gaussian distributed with unknown variances (i.e. the source powers), inspiring stochastic maximum likelihood DOA estimation. The DOAs of plane waves are estimated from multi-snapshot sensor array data using sparse Bayesian learning (SBL) where the noise is estimated across both sensors and snapshots. This SBL approach is more flexible and performs better than high-resolution methods since they cannot estimate the heteroscedastic noise process. An alternative to SBL is simple data normalization, whereby only the phase across the array is utilized. Simulations demonstrate that taking the heteroscedastic noise into account improves DOA estimation.

1:50

4pSP4. Multi-frequency sparse Bayesian learning for matched field processing in non-stationary noise. Kay L. Gemba, Santosh Nannuru, and Peter Gerstoft (MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu)

Using simulations and data, we localize a quiet source in the presence of an interferer. The SWellEx-96 Event S59 consists of a submerged source towed along an isobath over a 65 min duration with an interferer traversing the source track. This range independent, multi-frequency scenario includes mismatch, non-stationary noise, and operational uncertainty. Mismatch is defined as a misalignment between the actual source field observed at the array and the modeled replica vector. The noise process changes likely with time. This is modelled as a heteroscedastic Gaussian process, meaning that the noise variance is non-stationary across snapshots. Sparse Bayesian learning (SBL) has been applied previously to the matched field processing application [Gemba *et al*, J. Acoust. Soc. Am., 141:3411-3420, 2017]. Results demonstrate that SBL exhibits desirable robustness properties and improved localization performance when compared to the white noise constraint and Bartlett processors.

2:10

4pSP5. Passive acoustic detection of surface ships at ranges exceeding 100 kilometers and mechanisms for ship noise generation. Matthew E. Schinault (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, schinault.m@husky.neu.edu), Heriberto A. Garcia (Elec. and Comput. Eng., Northeastern Univ., Arlington, MA), Chenyang Zhu, Anna Kaplan, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

The passive ocean acoustic waveguide remote sensing technique is employed to detect diesel-electric vehicles at ranges exceeding 100 kilometers. The underwater sounds radiated from these vessels are received at long ranges on a large-aperture densely-sampled horizontal coherent hydrophone array. The source levels of these signals are estimated by correcting the received pressure levels for transmission losses modeled using a calibrated parabolic equation-based acoustic propagation model for random range-dependent ocean waveguides. Here we find spectra of ship-radiated sound that is extremely dynamic containing both broadband signals and narrowband tonals at discrete frequencies with source levels that vary depending on ship conditions. We track a vessel with increasing range to find range dependence on broadband signals at close range and tonal signals at long range. Machinery noise generated from engines, propellers, flow noise and other cavitation sources are found to vary depending on ship conditions and are unique to each vessel. Our analysis indicates these vessels can be instantaneously tracked over wide areas spanning more than 300 kilometers in diameter.

Contributed Papers

2:30

4pSP6. Comparison of adaptive and compressed sensing beamformers. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

We will review the capabilities and requirements of the Minimum Variance Distortionless Response (MVDR) and sparsity-seeking beamformers, from a system design perspective. We will focus on the problem of detecting a quiet source in the presence of loud interferers. These methods are solutions to particular and different optimization problems, requiring particular combinations of inputs. Thus, for example, MVDR does not require us to model the interference, whereas the sparsity-seeking methods need both interferer and quiet source models to be put into their dictionary. MVDR is notorious for requiring more precise calibration than is needed for conventional beamformers. We will present demonstrations on simulated and experiment data, illustrating key differences between the two methods.

2:45–3:05 Break

3:05

4pSP7. Extraction of shaft frequency based on the DEMON line spectrum. Mingyu Song, JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Systems Eng. Res. Inst. of CSSC, Beijing CuiWei St., Beijing, China, 56608812@qq.com), and Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Target-identification is an important technique in modern sea war. As the core device to identify targets, negative sonar analyzes noise underwater and abstracts their properties and determines its source. Shaft frequency which barely lies in the properties of the propeller differs among different kinds of ships, so this property can be regarded as a key to identify targets. This paper studied the theory and math model of the noise produced by ships. And Detection of Envelope Modulation on Noise (DEMON) is an effective technique to abstract low-frequency properties from high-frequency signals via demodulation. There are two methods to demodulate the noise, namely absolute low pass demodulation and square low pass demodulation. To improve the performance of DEMON spectrum estimation, adaptive line spectrum enhancement technology is introduced. An adaptive threshold is then set to eliminate the continuous spectrum. Then the largest common divisor algorithm is presented to calculate to shaft frequency out of the array consist of the centre frequency of these line spectrum. Simulation is conducted as described above. Two methods of demodulation is compared in this part. Brief sea experiment data processing shows that this algorithm is able to abstract the shaft frequency among the ship noise.

3:20

4pSP8. Research on suppression for tow-ship interference. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn), Mingyu Song, and JiangQiao Li (Systems Eng. Res. Inst. of CSSC, Beijing, China)

Towing line array underwater detection equipment detection performance will be tug of radiation noise interference, in order to enhance the performance of sonar detection, the key from the array spacing and follow-up signal processing to proceed. The purpose of using the towing array underwater probing device is to target the tail of the tug, but in actual use the towed array of half-wave lengths is affected by the strong tug noise The detection performance of the ship's tail direction is greatly restricted, and it is imperative to improve it. Otherwise, it will not be able to detect the threat target from the wake of the wake, and the target can not be found in time. This paper focuses on the reception directivity of the quarter-wave spacing matrix under the single-frequency signal and the wideband signal. By comparing the results of the two directional simulations, it is known that, unlike the half-wavelength interval matrix, the end-to-receive directionality of the quarter-wave spacing matrix is asymmetric, which can significantly suppress the interference in the direction of the drag Detection performance. The tandem noise can be suppressed by the combination of the received signal and the zeroing of the tampon, and the quarter-wave spacing matrix formed by the permutation-weighted combination array can further suppress the tug noise.

Invited Paper

3:35

4pSP9. Estimating the velocity of a moving acoustic source based on chirplet transform. Ningning Liang, Yixin Yang, Xijing Guo, and Boxuan Zhang (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, liangningning96@gmail.com)

The instantaneous frequency (IF) of a pure tone emitted by a transiting acoustical source varies due to the acoustical Doppler effect, which is usually used to estimate the velocity of the source. Hence, the common practice is first to estimate the IFs by the time-frequency analysis (TFA) of the signals and then to derive the velocity from the IFs. In this paper, a TFA method based on the chirplet transform is proposed where the IFs of the chirplets are formulated by the nonlinear kinetic function with respect to the source velocity. By iteratively fitting the chirplets to the signals it directly renders the velocity of the source. The efficiency of the proposed method is validated by the microphone recorded noise data due to a flying helicopter during an experiment carried out in May, 2018, northeast to the coast of Sanya, Hainan province, China.

Contributed Papers

3:55

4pSP10. The frequency choice of continuous-wave signal in low frequency active detection. Yiming Gu, Yanjun Zhang, ZaiXiao Gong, and Zhenglin Li (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring West Rd., Haidian District, Beijing, Beijing 100190, China, guyiming16@mails.ucas.ac.cn)

In active sonar, continuous-wave (CW) pulses are used to evaluate the velocity of a target because of its high-resolution of velocity. Unfortunately, due to multipath interface in shallow water, the received CW signals are fluctuating. This phenomenon degrades the performance of velocity measurement. The interference structure of low frequency sound field in shallow water is relatively stable and the intensity striations are regular in range-frequency graph, which makes it possible to reduce the influence of amplitude fluctuations by reasonable frequency design. In this paper, we obtain the separations of striations caused by different modes interfering. According to the separations, appropriate frequencies of multi-frequency CW pulses were selected to compensate the fluctuations. The simulation and experimental results show that this method can mitigate the influence of frequency selective fading in shallow water channel.

4:10

4pSP11. Development of the application of the digital microphone array system for real time source localization. Gee-Pinn J. Too and Ke-Han Liao (Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan 70101, Taiwan, toojames@yahoo.com)

This research is to develop a digital microphone array system to build a real-time sound source localization system and to produce acoustical holography. Acoustical holography can be understood as taking a photo of sound which can be used to source localization. In the situation of free field with an unknown signal, source signal is restored by time reversal mirror(TRM) where a free-field impulse response function(IRF) is used to achieve the goal of real time source localization. During the processing, the unknown source location is identified by scanning over a possible space for source locations to find a maximum restored signal. In the present study, simulations and experiments of source localization are carried out in 2D and 3D space. In the experiment of source localization with single source, the estimation errors are below 10% in certain distances between source and microphone array. For two sources situation, the distance between two sources will limit the application of the present approach.

Session 4pUWa

Underwater Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Acoustic Vector Field Studies II

Kevin B. Smith, Cochair
Department of Physics, Naval Postgraduate School, 833 Dyer Rd., Bldg 232, Rm. 114, Monterey, CA 93943

Robert Barton, Cochair
NUWC, 1176 Howell St, Newport, RI 02841

Invited Papers

1:00

4pUWa1. Review of directional sensors for low frequency underwater acoustic applications. James A. McConnell and Timothy P. Rorick (Ultra Electronics - Undersea Sensor Systems, Inc., 4868 East Park 30 Dr., Columbia City, IN 46725, james.mcconnell@ultra-ussi.com)

In this review, we present some of the more common embodiments directional sensors take on for use in low-frequency underwater acoustic applications. Sensors that fit within this paradigm are typically substantially smaller than an acoustic wavelength and can form first- or second-order cardioid beams at frequencies well below 10 kHz. A mathematical treatment of scalar, vector, and dyadic acoustic fields is presented as it pertains to understanding the phenomenology the sensors are required to measure. Cardioid beam-forming and directivity index are explained to provide context of using directional sensors in aperture constrained situations. The electro-acoustic performance of various sensor concepts is presented using lumped parameter models. Packaging concepts for high-fidelity operation, low self-noise, and low electronic noise in the seawater environment are shown. Special problems such as using directional sensors in deep water or near impedance boundaries is also covered.

1:20

4pUWa2. The rate of energy transport and direct measurements of group velocity. Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu) and David R. Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Intensity Vector Autonomous Recorder (IVAR) is a bottom deployed system measuring both particle velocity and pressure (combined sensor). Results using IVAR in the Sediment Characterization Experiment (SCE17) conducted off New England (spring 2017), are presented. Data originate both from ship noise involving the closest point of approach (CPA) of a large transport vessel, and experimental SCE17 signals relating to Signal Underwater Sound (SUS) MK-64 explosive sources. Several vector and scaler metrics emerge based on different combinations of second-order acoustic fields; one is the rate of energy transport (U) to be emphasized here. For the CPA data, estimates of U versus frequency represent the speed of the net transport of acoustic energy [D'Spain *et al*., J. Acoust. Soc. Am. 89, 1991], and additional interpretations are presented. For the SUS data, arrivals from individual modes are resolved in time-frequency analysis. Vertical intensity is shown to be much less than horizontal, and estimates of U are identified with modal group velocity with some caveats to be discussed. For example, the group velocity associated with the Airy-phase for mode-2 (frequency ~29 Hz) is U ~ 1370 m/s. This estimate emerges directly from the analysis without recourse to range divided by travel time.

1:40

4pUWa3. Near field scattering measurements using acoustic vector sensors. Georges Dossot, Jessica A. Barker, Daniel Perez, and Robert Barton (NUWC, 1176 Howell St., Bldg 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil)

The objective of this research is to measure and characterize acoustic energy flow surrounding underwater objects in the near field condition. To accomplish this, we employ prototype acoustic vector sensors which simultaneously measure acoustic pressure and three-dimensional acoustic particle acceleration at a singular point, allowing for precise vector field measurements. In March of 2018, a test at the U.S. Navy's Dodge Pond Test Facility examined the intensity field surrounding hollow spheres at low *ka* (wavenumber × radius) values. Phase differences between varying acoustic path lengths from reflected, scattered, and creeping waves result in interference patterns around the object. Various intensity processing techniques are used to reconstruct the acoustic field. Instantaneous intensity describes the time-dependent energy flux of the field and can be divided into active (real) and reactive (imaginary) components, which represent physically real elements. Time-averaged intensity shows energy transport and can be visualized in the form of acoustic streamlines.

2:00

4pUWa4. Ambient vector field measurements in the Northwest Providence Channel. Thomas J. Deal, Derrick Custodio, and Benjamin Cray (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, thomas.deal@navy.mil)

Data collected by a low-frequency acoustic vector sensor in shallow water west of the Berry Islands, Bahamas, will be presented. The frequency- and direction-dependence of the ambient noise was used to identify potential sources, such as distant shipping, wind-driven surface noise and subsurface currents. Noise sources were validated using recorded Automatic Identification System (AIS) tracks, weather observations and Acoustic Doppler Current Profiler (ADCP) measurements. A model of wind-driven surface noise that accounts for propagation effects due to range-dependent bathymetry, sound speed profiles, and surface waves will also be presented and compared to measured data.

2:20–2:35 Break

Contributed Papers

2:35

4pUWa5. Vector field observations near the continental slope off Big Sur, California. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, kbsmith@nps.edu), Thomas J. Deal (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI), Paul Leary (Dept. of Phys., Naval Postgrad. School, Monterey, CA), Steven Seda, Benjamin Carpenter (U.S. Navy, Monterey, CA), and Fahmi Laksana (Indonesian Navy, Monterey, CA)

In this work, data collected approximately 3 km off the coast of Big Sur, California, by a low frequency vector sensor system will be analyzed. Directional and temporal variations in the ambient noise field will be evaluated, and causes will be considered including flow noise due to currents, surface (wind) noise, distant shipping, and marine mammals. Bearing estimation results for shipping will be compared with AIS tracks recorded during the period of deployment. Local sound speed measurements and bathymetry in the vicinity of the deployment area will be used as inputs to a two-dimensional propagation model that properly invokes reciprocity of the acoustic vector field. Surface wave spectra collected by a nearby buoy will be used to generate rough surface realizations in order to evaluate the impact of these perturbations on low frequency vector field structure. These results will be combined with acoustic data observations in attempts to determine the impact of propagation on the structure of the field. Such an approach may allow us to infer acoustic source levels of AIS traffic off the coast.

2:50

4pUWa6. Embedded real-time processing of acoustic vector sensors using a lightweight open-source microcontroller. Paul Leary (Phys., Naval Postgrad. School, 833 Dyer Rd., Spanagel Hall, Rm. 203, Monterey, CA 93943, pleary@nps.edu), Vladimir Dobrokhodov, Kevin Jones (Mech. and Aerosp. Eng., Naval Postgrad. School, Monterey, CA), and Kevin B. Smith (Phys., Naval Postgrad. School, Monterey, CA)

This abstract is intended for the special session “Acoustic Vector Field Studies.” In this work, we will show our progress towards a low-power, lightweight, embedded data acquisition system and real-time signal processor for acoustic vector sensors, using an open-source microcontroller. Modern advances in low-cost, low-power “systems-on-a-chip” (SoC) provide unprecedented computational power, and some state-of-the-art devices are particularly well designed for acoustic signal processing applications. Here, we present the results of performance experiments, demonstrating the speed and accuracy of sampling and processing audio-frequency signals for a wide variety of signals and sampling regimes. We demonstrate trade-offs between sampling rates and resolutions for accuracy and speed at desired frequencies, and how these may be dynamically managed by autonomous systems. Finally, we will share some of our work towards real-time acoustic beamforming for use on autonomous vehicles.

3:05

4pUWa7. Effect of acoustic horizontal refraction on DOA estimation with a single vector hydrophone. Jun Tang and Shengchun Piao (College of Underwater Acoust., Harbin Eng. Univ., Nantong St. 145, Harbin 150001, China, tangjun@hrbeu.edu.cn)

The effect of horizontal refraction (HR) on DOA estimation with a single vector hydrophone is studied. It has been demonstrated in a previous study that HR may bring about a significant deviation between the true bearing of the source and the direction of time-averaged sound flux at the receiver point, i.e., DOA estimation error. [J. Tang *et al*., 2018, 43(2), ACTA ACUSTICA (in Chinese)]. In the present work, the DOA estimation error in a more realistic scenario is studied, more precisely, source signals are considered to be wide-band instead of time-harmonic, and meanwhile ambient noise is added to the received signal. The ocean waveguide used in simulations is a 3D version of the standard ASA wedge, which implies that the HR considered in this work is raised from multiple reflections between a horizontal sea surface and a sloping sea bottom. The results of this work shall offer some reference to DOA estimation in circumstances with strong HR effects.

3:20

4pUWa8. Parabolic equation modeling of a seismic airgun array. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Richard L. Campbell (OASIS Inc., Lexington, MA)

The numerical modelling of seismic airgun sources, with a specific emphasis on the impact of sound on the marine ecosystem, is a challenging problem. In this paper, the simplified range-independent iso-velocity problem set up as part if the International Airgun Modelling Workshop (IAMW), is solved using the Parabolic Equation (PE). The acoustic pressure and particle acceleration are computed including the arrival time series and source energy level in deci-decade bands for ranges spanning from 30m to 30 km. The particle acceleration is computed by taking the spatial gradient of the pressure on the PE computational grid. The two significant challenges this problem poses to the PE solution are the very large bandwidth (5-4500 Hz) and the computational accuracy required for time-series and particle acceleration computations. Each of the metrics outlined in the statement of the problem are computed and presented. In addition to computing the required fields, the 17-gun volumetric array is placed in a 3D ocean environment in deep water in the Gulf of Mexico, with peak pressure computed out to 30km and the band integrated Signal Energy Level (SEL) computed to 400 km.

Session 4pUWb

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics: Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods II

Charles W. Holland, Cochair
Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804

Stan E. Dosso, Cochair
School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada

Chair's Introduction—1:00

Invited Papers

1:05

4pUWb1. Statistical inference approach applied to simultaneous vertical particle velocity and acoustic pressure measurements from SUS explosive sources made in the New England Mudpatch. David P. Knobles (KSA LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Peter H. Dahl, David R. Dall'Osto (Appl. Phys. Lab., The Univ. of Washington, Seattle, WA), and Preston S. Wilson (Mech. Eng. Dept. and the Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Pressure and particle velocity time series were measured in the New England "Mudpatch" in the Spring of 2017. The time series were generated from Signal Underwater Sound (SUS) MK-64 explosive sources and recorded by a bottom-mounted L-shaped array of hydrophones and a vector sensor system that recorded both pressure and particle velocity (1.3 m off the bottom) in a separate location. A maximum entropy approach was utilized to infer the marginal probability distributions of geo-acoustic parameter values of the seabed using both the particle velocity and the pressure. The seabed consists of a surface layer of mud over sand whose thicknesses are range- and azimuth dependent. The modeled vertical component of particle velocity is computed by a three-point derivative of the pressure fields generated by RAM-PE along the vertical axes centered at the depth of the vector sensor. The two-way travel time CHIRP data for the multiple range and azimuth-dependent sediment layers provide a constraint for the sediment thickness for each sound speed hypothesis of the model parameterization space. Results of the statistical inference of the geo-acoustic structure derived from the particle velocity measurements are compared to those derived from pressure field measurements. [Work supported by ONR.]

1:25

4pUWb2. Inferring low-frequency compressional sound speed and attenuation in muddy sediments from long-range broadband sound propagation. Lin Wan, Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, wan@udel.edu), David P. Knobles (Knobles Sci. and Anal., LLC, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX)

While researchers have successfully applied many geo-acoustic inversion methods, involving normal mode analysis or matched field processing, to invert seabed parameters in sandy bottoms, inferring low-frequency compressional sound speed and attenuation in mud is still a challenging problem, especially for a marine sediment where a mud layer overlies a sandy bottom. Recent studies show that there is an ambiguity between the low-frequency attenuation of mud and sand in three different inversion algorithms based on (1) modal amplitude, (2) transmission loss, and (3) spatial coherence measurements [JASA **143**, 1798 (2018)]. An increase (decrease) of mud attenuation can be compensated by decreasing (increasing) the attenuation in the sandy basement. In this paper, an inversion method combining these three inversion algorithms and statistical inference techniques (e.g., Bayesian-Maximum Entropy, [JASA **138**, 3563–3575 (2015)]) is utilized to analyze the long-range broadband acoustic signals measured by several vertical and horizontal line arrays during the Seabed Characterization Experiment 2017 conducted in the New England Mud Patch. Reliable estimates of the low-frequency mud properties are obtained by removing the ambiguity. Finally, the inverted results are compared with historical experimental data and theoretical predictions. [Work supported by ONR Ocean Acoustics.]

1:45

4pUWb3. In situ observation of sediment sound speed and attenuation from coarse to fine-grained sediments. Jie Yang (Acoust. Dept., APL-UW, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

The long-term objective is to address the impact of spatial variability of sediment properties on sound propagation and reverberation from a few hundred hertz to 10 kHz. In this band, the frequency dependences of sound speed and attenuation are important for applications, but cannot be inferred from high-frequency data, hence direct measurement is necessary. In this talk, a summary of *in situ* measurements of sediment sound speed and attenuation collected from three major field experiments, i.e., Shallow Water 2006 (SW06),

Target and Reverberation Experiment 2013 (TREX13), and Seabed Characterization Experiment 2017 (SCE17), are presented. Sediment sound speed and attenuation within the surficial 3 m of sediments were obtained through penetration, using the Sediment Acoustic-speed Measurement System (SAMS). In TREX13, *in situ* measurements were carried out at five sites along a 5-km track, with sediment types ranging from coarse sand to a mixture of soft mud over sand. SAMS was deployed at 18 sites within the 10×30 km SCE17 study area. Significant variation of sediment geoacoustic properties was observed in range (TREX13) and depth (SCE17), respectively. Preliminary sediment acoustic modeling work is presented, using SAMS data taken from the wide range of sediment types. [Work supported by ONR.]

2:05

4pUWb4. Bottom scattering by a moving, narrowband source. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu) and William S. Hodgkiss (UC San Diego, La Jolla, CA)

During the Target and Reverberation Experiment in 2013 (TREX13), two vertical line arrays were deployed and a mid-water source was towed past these arrays while transmitting 15-s long, mid-frequency CW tones. While the received signals exhibited the expected positive and negative Doppler shifts as the source moved toward and away from the arrays, a significant amount of energy was always present in the band between these two frequencies. This counterintuitive result is due to sound scattering from the seafloor in the vicinity of the source. This sound will experience a different Doppler shift and although it may be incident above the critical angle, it can scatter into propagating modes that can be received on the arrays. As opposed to wide-band reverberation where the energy received at a given time can be associated with an elliptical scattering patch, in this case the energy received at a given frequency can be associated with scattering from a hyperbolic scattering patch. A normal-mode reverberation model for this effect has been developed and is used to examine whether this data can be used to invert for the seafloor scattering strength at the TREX13 site. [Work supported by ONR.]

2:25–2:40 Break

Contributed Papers

2:40

4pUWb5. An experimental test of end-fire synthetic aperture sonar for sediment acoustics studies. Shannon-Morgan M. Steele and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping/Oceanogr., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, ssteele@ccom.unh.edu)

Seafloor sediment acoustic returns are comprised of scattering from both the interface and sediment volume. At low-frequencies, volume scattering is often the dominant mechanism; however, direct measurements of this component have rarely been made, due to interface roughness biasing caused by large beamwidths. End-fire synthetic aperture sonar (SAS) can achieve narrower beamwidths by coherently combining multiple acoustic returns as a vertically oriented transmitter and/or receiver is moved towards the seafloor. Beam pattern simulations suggest end-fire SAS can reduce the beamwidth of a sonar by a factor of 6 with an array length of 100 wavelengths. Achieving these gains is dependent on the ability to resolve relative sonar motion to at least an eighth of a wavelength. This talk will present results from an end-fire SAS field trial. Results will include an analysis of beamwidth gains achieved during the end-fire SAS field test and methods to improve these gains by using the scattered field to refine sonar positions.

2:55

4pUWb6. Studies on water column nutrient distribution and sound speed profile variations along coastal region of India. Sakthivel M. Santhanam (Electronics and Commun., SSN College of Eng., Rajiv Gandhi Salai, OMR Kalavakkam, Chennai, Tamil Nadu 603110, India, sakthivels@ssn.edu.in)

This study aim is to measure the water column nutrient and sound velocity variation along the coastal region of Poompuhar, Cuddalore, Mahabalipuram, Kalpakkam, Chennai and Pondicherry. Sound speed variation in the environment of the particular ocean depends on temperature, salinity, and pressure. Since temperature and salinity variations are large compared to pressure. In study locations the sound speed varies from 1445 to 1540m/s, temperature 27 to 29^0c and salinity 29 to 33psu. Water column nutrients like Nitrate, Nitrite, Phosphate, Silicate and Urea were analyzed to understand the physical and chemical concentration of study location. This nutrient has an inverse relationship with the temperature. Location-based nutrients analysis has been carried to identify the concentration each nutrients level in the particular location. The maximum concentration of nutrients observed in the study locations are silicate in Pondicherry (84%), Nitrite in Kalpakkam (4%), Nitrate in Cuddalore (16%), Phosphate in Chennai (6%), and Urea Poompuhar-2 (24%). The total suspended solids level in the water column is measured to identify the impact on the signal transmission in underwater. The higher value of TSS is observed in Cuddalore (0.15 mg/L).

3:10

4pUWb7. Measurements of modal attenuation using broadband sources in the New England Mud Patch. Kerry Unrein, Gopu R. Potty, James H. Miller (The Univ. of Rhode Island, Narragansett, RI 02882, kerrycutler@my.uri.edu), Dag Tollefsen (Norwegian Defense Res. Establishment, Horten, Norway), David P. Knobles (KSA, LLC, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX), and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Measurements of underwater acoustic signals were made on a bottom-mounted horizontal line array during the Seabed Characterization Experiment (SCEx) in the New England Mud Patch south of Martha's Vineyard in about 70 meters of water. The signals were generated by SUS (Signals, Underwater Sound) charges detonated at various locations in the experimental area at a depth of 18 m, during nearly-isovelocity conditions. The broadband signals were analyzed for modal arrival time and amplitude using time-frequency techniques. Ratios of modal amplitudes at the different hydrophones were used to estimate the modal attenuation coefficients. Hence, these estimates are independent of any uncertainty in the frequency-dependent source level of the SUS charges. These coefficients are directly related to the depth-dependent sediment attenuation profile. A sensitivity study was performed to understand how the modes sample the different layers and provides estimates of resolution kernels. Posteriori error analysis provides averages and standard deviations for the estimate of sediment attenuation as function of depth. The frequency bands of interest range from 10 Hz to 200 Hz for modes 1 to 4. We will compare our estimates of sediment attenuation with historical measurements. [Work supported by Office of Naval Research.]

3:25

4pUWb8. On the use of conventional, adaptive beamforming and the Fourier-Transform-based algorithm for estimating seabed properties.
Lanfranco Muzi, Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu), and John Gebbie (Metron, Inc, Portland, OR)

Knowledge of the bottom reflection coefficient as a function of frequency and grazing angle can be used to infer seabed properties. A known passive technique for estimating the bottom reflection coefficient is based on conventional beamforming of natural marine ambient noise over a vertical line array. In this study, the extension of the technique to adaptive beamforming is formally investigated, and compared to a more recent Fourier-Transform-based algorithm. This is done by developing both a mathematical formalization of the high-resolution algorithm in the discrete space-wavenumber domain, and a mathematical proof of the original technique (based on conventional beamforming). It is shown that replacing conventional beamforming with adaptive beamforming cannot be guaranteed to provide an estimate of the bottom reflection coefficient. The conclusions are demonstrated on simulated and measured data. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

OPEN MEETINGS OF TECHNICAL COMMITTEES/SPECIALTY GROUPS

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, 6 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Rattenbury A/B (FE)
Acoustical Oceanography	7:30 p.m.	Esquimalt (VCC)
Animal Bioacoustics	7:30 p.m.	Oak Bay 1/2 (VCC)
Architectural Acoustics	7:30 p.m.	Theater (VCC)
Physical Acoustics	7:30 p.m.	Colwood 1/2 (VCC)
Psychological and Physiological Acoustics	7:30 p.m.	Salon B (VCC)
Speech Communication	7:30 p.m.	Salon A (VCC)
Structural Acoustics and Vibration	7:30 p.m.	Saanich 1/2 (VCC)

Committees meeting on Wednesday, 7 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Sidney 1/2 (FE)
Signal Processing in Acoustics	7:30 p.m.	Colwood 1/2 (FE)

Committees meeting on Thursday, 8 November

Committee	Start Time	Room
Computational Acoustics	4:30 p.m.	Esquimalt (VCC)
Musical Acoustics	7:30 p.m.	Crystal Ballroom (FE)
Noise	7:30 p.m.	Shaughnessy (FE)
Underwater Acoustics	7:30 p.m.	Rattenbury A/B (FE)

Session 5aAA

Architectural Acoustics and Noise: Session in Memory of Murray Hodgson I

Nicola Prodi, Cochair
Dept. of Engineering, University of Ferrara, via Saragat 1, Ferrara 44122, Italy

Maureen R. Connelly, Cochair
Construction and Environment, BCIT, 3700 Willingdon Ave., Building NE03 Office 107, Vancouver, BC V5G3H2, Canada

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Chair's Introduction—8:30

Invited Papers

8:35

5aAA1. Murray Hodgson: An appreciation from a practicing acoustical consultant. John P. O'Keefe (O'Keefe Acoust., 10 Ridley Gardens, Toronto, ON M6R 2T8, Canada, john@okeefeacoustics.com)

So much of what we know about the behaviour of sound in rooms comes from studies in concert halls and reverberation rooms. Hardly everyday architectural geometry. The author first met Murray Hodgson in Cambridge when Dr. Hodgson was doing his early scale model studies on factories. Geometrically and acoustically, factories are much more akin to the day to day built environment that we inhabit. The geometry of a factory is typically long, wide and very flat with scattering elements, typically on the floor. Murray would go on to apply his work on factories to other long, low, and wide rooms. Our understanding of the acoustics of open plan offices, health care facilities, and, of course, classrooms can be traced back to his post-doctoral work in Cambridge. One is impressed how the work on factory acoustics grew to cover so much of the rooms we live in. More impressive was his ability to tackle questions that others wouldn't. Questions that acoustical consultants are often asked and really don't have an answer for yet. Noise control in naturally ventilated buildings for example. His legacy will show this work as seminal in the nascent green building type genre.

8:55

5aAA2. An overview of Murray's Hodgson indirect contribution to the acoustical community as a mentor. Banda Logawa (BKL Consultants, 308-1200 Lynn Valley Rd., North Vancouver, BC V7J 2A2, Bangladesh, logawa@bkl.ca)

During his more than 20 years of career as a professor at University of British Columbia, Murray Hodgson has inspired many of his previous students to pursue their career in various fields of acoustics, both in academia and industry. His undergraduate engineering course Acoustics and Noise Control was the first exposure to the vast world of acoustics for many of his former students. Furthermore, Murray has also supervised many students through their graduate studies. This paper will highlight Murray's indirect contribution as a mentor by summarizing the current whereabouts of his former students who are now an active member of the acoustical community in North America and around the world.

9:15

5aAA3. Some contributions of Murray Hodgson to room acoustics modeling. Vincent Valeau (Institut PPRIME UPR 3346, CNRS-Université de Poitiers-ENSMA, 6 rue Marcel Doré TSA 41105, Poitiers Cedex 9 86073, France, vincent.valeau@univ-poitiers.fr), Judicaël Picaut (LUNAM Université, IFSTTAR, AME, LAE, Bouguenais Cedex, France), and Cedric Foy (CEREMA, Laboratoire Regional de Strasbourg, Strasbourg, France)

This presentation focuses on some of the many contributions of Murray Hodgson (1952–2017) concerning room-acoustic modeling. The concepts of sound field diffuseness and diffuse reflections have always been of major interest for Murray, his studies associating most of the time numerical modeling and measurements on scale models or real-scale rooms. His most cited paper [JASA 89, 1990] proposes to include diffuse surface reflections in ray-tracing simulations, and is a milestone in room acoustics prediction as most standard room-acoustic prediction softwares now include surface diffusivity. Another extensively cited paper [App. Ac. 49, 1996] discusses the applicability of diffuse field theory according to room shape and to absorption distribution and magnitude. The conclusions of this study are still greatly useful to researchers, students, and practitioners. Another important focus of Murray's research has been the acoustics of « fitted » rooms (i.e., rooms containing many obstacles such as industrial workrooms, classrooms…), and was the topic of many of his papers. This presentation will review some contributions of Murray on this research topic.

9:35

5aAA4. Diffuseness. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

In architectural acoustics and noise control, the diffuse sound field is the fundamental assumption. The key of its definition is isotropy and incoherence, thus infinitely many sound uncorrelated waves arrive from directions uniformly distributed over a sphere. The room shape and the amount and placement of absorbing and scattering surfaces and objects determine the isotropy of actual sound fields in rooms. Spatial and directional distributions of room sound fields are reviewed and discussed concerning the consequences on decay curves and on the homogeneity of sound energy density in rooms. Several references to Murray Hodgson illustrate his contributions to this field, particularly to applications to noise control in workrooms.

9:55–10:10 Break

10:10

5aAA5. Reflections on the "Topical Meeting on Classroom Acoustics" at the Acoustical Society of America's spring 2005 meeting in Vancouver. 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)

I had the pleasure of co-organizing the "Topical Meeting on Classroom Acoustics" at the Acoustical Society of America's spring 2005 meeting in Vancouver with Murray Hodgson. The topical meeting featured 26 presentations from a diverse group of international scholars; Hodgson and his research team presented seven of those. This talk will include my personal reflections from co-organizing the session with Hodgson, but also discuss the impact of the work presented by Hodgson and his students at that topical meeting. Special attention will be given to how the talks given by Hodgson and his research team demonstrate the breadth of his varied interests in classroom acoustics and point to the significant accomplishments he made in this area throughout his career. Hodgson's body of work continues to heavily influence my own and many others who study classroom acoustics.

10:30

5aAA6. Do we still need diffuse field theory? Francesco Martellotta (DICAR, Politecnico di Bari, Via Orabona 4, Bari, Bari 70125, Italy, francesco.martellotta@poliba.it)

More than twenty years after Murray Hodgson's "When is diffuse field theory applicable?" paper, we gathered more and more evidences that diffuse field is mostly a chimera. If we consider the two most important implications of the diffuse field model, i.e., sound pressure level uniform distribution and reverberation time invariance, it is quite easy to say that, based on actual measurements in a number of different spaces, such conditions are hardly found. Ideal sound diffusion requires ergodic and mixing conditions, which are not obvious to happen, particularly when sound absorption is unevenly distributed or rooms are not proportionate. So, apparently, diffuse field theory could even be dismissed in favour of more accurate approaches capable of taking into account the specific nature of each space. Nowadays, we have several instruments spanning from the many variations of the ray-tracing algorithm to the "brute force" numerical solution of the wave equation. However, such methods rely on the measurement or estimation of other coefficients that, if not properly made, may introduce even bigger inaccuracies. A critical analysis is carried out showing that diffuse field theory still represents an important way to understand sound propagation in enclosed spaces.

10:50

5aAA7. Concave surfaces and acoustics of performance spaces—Part I— Hybrid ray-image analysis. Eva M. Johnston-Iafelice and Ramani Ramakrishnan (Bldg. Sci., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, ejohnstoniafelice@ryerson.ca)

Current acoustic practices deem that concave surfaces do not provide good acoustical performance. However, old cathedrals, churches, and enclosed performance spaces with concave interiors seem to perform well. Part I of the current investigation analyzes the acoustical performance of spaces with curved surfaces. The main focus of the current investigation was to research the uniformity of the sound field produced by curved surfaces by analyzing sound pressure level distribution throughout the audience space. It studied the impact of the focal plane on the overall sound distribution within an enclosed space. To analyze the effect of curved surfaces at different frequencies, three enclosed rooms with curved surfaces were used to measure the sound pressure levels throughout an audience space: the Paul Crocker Gallery in the Ryerson Architecture Building, Toronto; St. Martin-in-the-fields Anglican Church, Toronto; and Wigmore Hall, United Kingdom. The evaluations were achieved with both experimental methods, and computer simulations using hybrid-ray-image methods. Computer simulations were validated by the initial on-site measurements in the Toronto locations. After these evaluations were performed, results showed that in these conditions, the curved surfaces had minimal negative impact as perceived by the audience. The results of the investigation will be presented in this paper.

11:10

5aAA8. Concave surfaces and acoustics of performance spaces—Part II —Wave analysis. Ramani Ramakrishnan and Eva M. Johnston-Iafelice (Architectural Sci., Ryerson Univ., 325 Victoria St., Toronto, ON M5B 2K3, Canada, rramakri@ryerson.ca)

Conventional wisdom states that having concave surfaces as the envelope of any occupied space does not produce good sound. The focussing effect of concave surfaces can cause high sound pressure levels, coloration, and echoes. However, throughout history there have been many enclosed rooms with large curved surfaces as envelopes that seem to produce good acoustics. Recent research suggested that wave analysis must be undertaken to establish the impact of concave surfaces. In contrast to Part I of the current investigation, evaluation of the sound pressure level distribution, in rooms with concave surfaces, was performed by solving the governing wave equation. The main reason is that the image-ray theory is valid only at frequencies greater than the Schroeder cut-off frequency. The wave theory is used for frequencies lower than 100 Hz. Finite element modelling was applied to solve for the sound pressure level distribution within

rooms with concave surfaces. Three spaces, the Paul Crocker Gallery in Ryerson University, Toronto, St. Pauls Anglican Church in Toronto and Wigmore Hall in London were investigated in this study. The results for three low frequencies (25 Hz, 50 Hz, and 100 Hz) as well as their combination will be presented in this paper.

FRIDAY MORNING, 9 NOVEMBER 2018 SHAUGHNESSY (FE), 8:45 A.M. TO 12:00 NOON

Session 5aAB

Animal Bioacoustics: Marine Mammal Bioacoustics I

Susanna B. Blackwell, Chair
Greeneridge Sciences, Inc., 120 Tamarack Drive, Aptos, CA 95003

Contributed Papers

8:45

5aAB1. Clymene dolphin (*Stenella clymene*) whistles in the Southwest Atlantic Ocean. Juliana R. Moron (Instituto Aqualie, Av. Doutor Paulo Japiassu Coelho, n° 714 sala 202, Juiz de Fora 36033310, Brazil, juliana-moron@hotmail.com), Luiz Cláudio P. Alves, Carla Viviane de Assis, Felipe C. Garcia (NAV Oceanografia Ambiental, Rio de Janeiro, Rio de Janeiro, Brazil), and Artur Andriolo (Laboratório de Bioacústica e Ecologia Comportamental - Instituto de Ciências Biológicas, Universidade Federal de Juiz de Fora, Juiz de Fora, Minas Gerais, Brazil)

The Clymene dolphin despite being endemic for the Atlantic Ocean continues to be the least known species in the genus *Stenella* without available information on their vocal repertoire in Brazilian waters. Data were obtained during mitigation and monitoring work required by IBAMA under the federal environmental licensing as conditions of the license 108/16 for the 3D seismic survey in Pará/Maranhão Sedimentary Basin process 02022.000015/2014. The species record was performed onboard the vessel Polarcus Alima with a Mseis (Night Hawk III) four-element towed array passing signals to a digital M-Audio, recording at 96 kHz/16bits. During visual and acoustic monitoring with the air-guns off, a group of approximate 80 dolphins were sighted and recorded on May 8th, 2016 (00°38’00” N, 44°45’25” O) at 3,425 m depth. The wav-files were analyzed through the spectrogram configured as DFT 2048 samples, 70% overlap and Hamming window of 1024 points generated by software Raven Pro 1.5. The results of 14 min recording allowed the extraction of 257 whistles. Minimum frequencies ranged from 5.20 kHz to 15.92 kHz (mean = 10.99 kHz); maximum frequencies ranged from 7.72 kHz to 28.04 kHz (mean = 16.16 kHz); whistle duration ranged from 0.01 s to 1.52 s (mean = 0.43 s). These results are important since they represent the first acoustic record of Clymene dolphins in Brazil.

9:00

5aAB2. Burst-pulses in East Greenland narwhals: Further evidence for unique, individual-specific vocalizations. Susanna B. Blackwell (Greeneridge Sci., Inc., 120 Tamarack Dr., Aptos, CA 95003, susanna@greeneridge.com), Outi M. Tervo (Greenland Representation, Greenland Inst. of Natural Resources, Copenhagen, Denmark), Alexander S. Conrad (Greeneridge Sci., Inc., Santa Barbara, CA), and Mads P. Heide-Jørgensen (Greenland Representation, Greenland Inst. of Natural Resources, Copenhagen, Denmark)

In August 2013–2017, narwhals from Scoresby Sound, East Greenland, were instrumented with satellite-linked transmitters and acoustic sound and movement tags (Acousonde™) over periods of up to 8 days. The records obtained provided continuous information on the whales’ acoustic behavior during foraging and social interactions. Burst-pulses were the most easily recognizable non-feeding vocalization. They tended to occur near the surface and were more common in records that displayed higher rates of clicks from other individuals, i.e., records that presumably would include more social interactions with other narwhals. Over 100 burst pulses detected in eight subjects were analyzed in terms of their overall length and their pattern of successive inter-click intervals (ICIs). While overall length could be somewhat variable, the succession of ICIs was unique for each whale and could therefore have an identity-carrying function, as the signature whistle does in bottlenose dolphins and other odontocetes. The occurrence of these individual-specific burst-pulses in time and space will be examined in relation to other factors provided by the tags, such as the time of day, whale depth, presence of conspecifics, and behavioral state. [Work sponsored in part by the Greenland Institute of Natural Resources.]

9:15

5aAB3. Vocalizations of common minke whales (*Balaenoptera acutorostrata*) in an eastern North Pacific feeding ground. Jared Towers (Bay Cetology, Alert Bay, BC, Canada) and Katrina Nikolich (Biology, Univ. of Victoria, 992 Karen Crescent, Victoria, BC V8X3C6, Canada, katrinan@uvic.ca)

The minke whale (*Balaenoptera acutorostrata*) is the smallest species of baleen whale and has a cosmopolitan distribution. Despite extensive study on the vocalizations of other balaenopterids, the acoustic repertoire of minke whales is not well known. Individuals of the North Pacific subspecies of common minke whale (*B. a. scammoni*) are known to produce unique vocalizations ("boings") during their putative breeding season from fall to spring. However, no vocalizations have been previously reported for this subspecies in summer feeding grounds. We present four novel call types recorded in the presence of minke whales in Cormorant Channel, in coastal British Columbia, Canada, during the summer of 2012. These calls consist of broadband pulses, tonal wavers, downsweeps, and pulse trains. Calling rates for minke whales in this study region were very low compared to those reported for North Atlantic minke whales on their feeding grounds. We compare our candidate call types with vocalizations described for other minke whale populations and propose predation risk as a cause of the low calling rates observed for minke whales in Cormorant Channel.

9:30

5aAB4. Temporal variations in humpback whale (*Megaptera novaeangliae*) song in Monterey Bay National Marine Sanctuary, northeast Pacific. John P. Ryan (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, ryjo@mbari.org), Danelle E. Cline (Monterey Bay Aquarium Res. Inst., Moss Landing, CA), John E. Joseph (Naval Postgrad. School, Monterey, CA), Tetyana Margolina (Naval Postgrad. School, Monterey, CA), Alison K. Stimpert (Moss Landing Marine Labs., Moss Landing, CA), Karin Forney (NOAA, Moss Landing, CA), Nancy Black (Monterey Bay Whale Watch, Monterey, CA), Andrew DeVogelaere (NOAA, Moss Landing, CA), Mark Fischer (Aguasonic Acoust., Rio Vista, CA), Christopher Wahl, and Francisco Chavez (Monterey Bay Aquarium Res. Inst., Moss Landing, CA)

Using two years of nearly continuous recordings from Monterey Bay National Marine Sanctuary, August 2015 through July 2017, variations in humpback whale song are examined on diel, seasonal, and interannual time scales. The cabled hydrophone is in humpback feeding and migratory habitat at 36.7128°N, 122.186°W. Diel analyses show 69% of song during night, 23% during day, and 8% during dusk or dawn. Seasonal analyses show song absence during summer (June–August), emergence during fall (September–October), peak during late fall/winter (November–January), and highly variable detection during spring (February–May). During both years >80% of song occurred during the November–January peak. Song detection within a month reached a maximum of 58% of the time during November 2016. Song length increased ($p < 0.01$) month-to-month from the start in fall through the end of the peak in January. The months of maximum song occurrence coincide with declining visual sighting of humpbacks within Monterey Bay, consistent with seasonal southward migration to breeding habitat and/or more offshore residence. Interannual variation in song was marked by a 51% increase between the first and second years. Habitat during the first year was strongly influenced by a prolonged regional oceanic warm anomaly and the largest toxic algal bloom ever recorded in the northeast Pacific. Alternative hypotheses for the interannual difference in song detection relate to differences in humpback regional abundance, behavior, and habitat occupancy.

9:45

5aAB5. What the African naked mole-rat can tell us about mammalian hearing and acoustic communication. Catherine Barone (Anatomy & Cell Biology, Univ. of Illinois Chicago, 808 S. Wood St., Rm. 578 MC 512, Chicago, IL 60612-7308, cbaron2@uic.edu), Thomas Park (Biological Sci., Univ. of Illinois Chicago, Chicago, IL), and Sonja Pyott (Univ. medical Ctr. Groningen, Groningen, Netherlands)

African naked mole-rats are eusocial rodents that live underground in narrow burrows. Likely reflecting adaptations to their subterranean environment, naked mole-rats (NMRs) have poor high frequency hearing, relatively high auditory thresholds, and limited ability to localize sound thus providing a unique opportunity to investigate the mechanisms required for high frequency hearing and sound localization. Despite their poor hearing sensitivity, NMRs display a functional coupled auditory-vocal communication system. Prior work revealed they use 17 different vocalizations to communicate within their colonies. Due to their unusual hearing, investigation of NMRs, especially in comparison to other rodent models, provides a unique opportunity to identify features of the auditory system that underlie high frequency hearing, sound localization, and discrimination of vocalizations. Our recent anatomical characterization indicates the NMR cochlea undergoes stunted maturation of peripheral synaptic organization and altered expression of ion channels compared to other mammals. Our more recent function auditory assessments indicate multiple mechanisms that may result in their poor hearing. Overall, these studies provide a framework for comparative insight into the peripheral mechanisms required for high frequency hearing and sound localization.

10:00

5aAB6. Do singing humpback whales (*Megaptera novaengliae*) favor specific frequency bands? Christina E. Perazio (Evolution, Ecology, and Behavior, State Univ. of New York at Buffalo, Buffalo, NY, cperazio@buffalo.edu) and Eduardo Mercado (Psych., State Univ. of New York at Buffalo, Buffalo, NY)

Many species alter their acoustic signals in response to environmental or anthropogenic factors. Differential use of frequency bands may occur as a result of overlapping biotic and abiotic sounds that interfere with communication signals between conspecifics. Singing humpback whales produce units with frequencies ranging from 30 Hz to over 10,000 Hz, but may favor specific frequency bands during song production. We tested the prediction that singing humpback whales consistently produce units with energy focused in specific spectral bands by measuring the consistency of peak frequency bands in the southeastern Pacific stock G population of humpback whales in the Gulf of Tribugá, Colombian Pacific. Preliminary results document extensive use of units with peak frequencies clustered in two different bands between 300 Hz and 1000 Hz for the dominant theme, and additional peak frequency bands between 300 Hz and 3000 Hz for other song themes. These analyses will allow for further spatial and temporal comparisons across distinct population segments, and can be used in future studies to assess whether singers change their use of frequency bands in response to increased anthropogenic noise or as themes evolve over time.

10:15–10:30 Break

10:30

5aAB7. Acute motor and vocal response of humpback whales (*Megaptera novaeangliae*) to playback of amplitude-modulated noise: A method to test frequency range of hearing. Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Mark H. H. Deakos (HDR Inc., Lahaina, HI), Keith Jenkins (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), Brian C. Balmer (National Marine Mammal Foundation, San Diego, CA), Alan P. Ness (HDR Inc., Lahaina, HI), and Rachel Cartwright (California State Univ. Channel Islands, Camarillo, CA)

The frequency hearing range of mysticete cetaceans remains elusive despite a growing concern about the negative impacts of anthropogenic noise. A playback study was conducted in the Maui Nui region of Hawaii during March 2018. Stationary humpback whales (i.e., single singers or silent dyads) were targeted due to their predictable behavior that allowed for observations pre, during, and post-playback. Playback consisted of amplitude-modulated noise that was broadband, high-pass filtered, or 1/3 octave bandpass filtered. The animal's motor behavior was recorded by video from a swimmer positioned above the animal. Acoustic playback and acquisition was performed from a small vessel approximately 41 m to 56 m from a targeted animal. A total of 16 playbacks were conducted with estimated received levels between 116 dB–138 dB RMS (re 1 μPa). Motor responses ranged from abruptly swimming away to subtle fluke movements with a median response latency of 1.7 sec (n = 7). Vocal responses ranged from abrupt song termination to unexpected changes in song unit pattern, with a median response latency of 2.94 sec (n = 8). This novel playback methodology was effective at producing and measuring motor and vocal responses to sound and may be useful in measuring the hearing range of mysticetes.

10:45

5aAB8. Mapping the phonetic structure of humpback whale song units. Howard S. Pines (Retired, Wireless Network Business Unit, Cisco Systems, 8752 Terrace Dr., El Cerrito, CA 94530, howardpines@gmail.com)

The striking similarities of time-frequency spectrograms of voiced human speech and humpback whale vocalizations suggested a common targeted frequency-modulated phonetic basis. To map the sub-unit structure of humpback whale song units, a time-frequency contour segmentation, extraction, and classification procedure was developed and tested on streaming voiced human speech. When the procedure was applied to humpback vocalizations and the tone-pairs of the two most energetic "vocal fold" harmonic frequencies were plotted in x-y coordinates, the plot exhibited properties of an optimally structured Shannon "modem symbol constellation" diagram of 14 distinct sub-regions and 60 acoustically distinct sub-unit symbols. The humpback symbol constellation is structurally comparable to the tone-pair symbol constellations of English and Asian language vowels. The information entropy and plot of the humpback sub-unit symbol set's cumulative

probability vs. ranked frequency distribution function are nearly identical to the entropy and Zipf power law profile of the English language phoneme set. The precise specification of the sub-unit structure of more than one hundred song units analyzed to date has resulted in a three-fold expansion in the number of unique units identified in previous studies, suggesting that the "lexicon" of humpback song units is potentially much larger than cited in the literature.

11:00

5aAB9. Airborne and underwater acoustic repertoire of hooded seals (*Cystophora cristata*): Cornerstone for acoustic monitoring. Heloise Frouin-Mouy (JASCO Appl. Sci., 2305 - 4464 Markham St., Victoria, BC V8Z 7X8, Canada, heloise.frouin-mouy@jasco.com)

The hooded seal is a migratory species inhabiting the North Atlantic. They whelp and breed during mid- to late March on pack ice near Jan Mayen Island, in the Davis Strait, off the northeastern Newfoundland coast, and in the Gulf of St. Lawrence. After breeding, hooded seals return to the pack ice off eastern Greenland to moult during June and July, and then they disperse broadly for summer and fall before returning to their respective breeding areas. Passive acoustic monitoring conducted over spatial scales consistent with known and potential habitat of the hooded seal could add insight into seasonal, diel, and spatial occurrence patterns of this species. To better characterize its acoustic repertoire (notably underwater calls), airborne and underwater acoustic signals of hooded seals were recorded during their breeding season on the pack ice in the Gulf of St. Lawrence from 12 to 17 March 2018. Hood and septum noises were the predominant sounds heard from males on the ice surface. Hooded seal underwater acoustic repertoire is larger and more diverse than has been previously described. Using the dataset from an extended acoustic monitoring program along Canada's East Coast, some clues are provided about the seasonal distribution of hooded seals.

11:15

5aAB10. Vocalizations of North American river otters (*Lontra canadensis*) in two human care populations. Sarah Walkley (Univ. of Southern MS, 254 Kitchawan Rd., South Salem, NY 10590-2014, sarahnwalkley@gmail.com), Maria Zapetis (Univ. of Southern MS, San Diego, CA), and Heidi Lyn (Univ. of Southern MS, Gulfport, MS)

There is a dearth of information regarding the vocal repertoire of North American river otters (Lontra canadensis). This indicator species is cosmopolitan yet elusive, making recordings methodologically difficult in the wild. Therefore, this exploratory study uses video and audio recordings of two populations of North American river otters in human care to broaden the known vocal repertoire of river otters in various social contexts. The populations consist of a male-female and a male-male pair. This study is the first to examine the vocalizations produced in a male-male pair of river otters. Call types were acoustically distinguished based on their appearance on a spectrogram. Parameters including average duration, frequency (high, low, max, 1st quarter, center, and 3rd quarter), and power (max and average) were measured for each call. Because vocalizations are the focal point of this study, only behaviors co-occurring with vocalizations were included in the chi square analysis that showed a significant relationship between call type and behavior. Squeaks and whines were present during agonistic behaviors while chirps were produced during non-agonistic behaviors including investigating, stationary, and grooming. Results support that behavior likely plays a role in the type of calls produced by river otters in human care.

11:30

5aAB11. Temporal separation in call types found for large baleen whale species in offshore waters of the Canadian Pacific. Rianna Burnham (Univ. of Victoria, Victoria, BC V8P 5C2, Canada, burnhamr@uvic.ca)

Populations of large whale species were severely reduced by commercial whaling. Studies of repopulation and habitat use are hindered by their use of offshore waters. Passive acoustic recordings made using stationary and mobile receivers in in- and offshore waters off the west coast of Vancouver Island are used to begin to re-establish abundance and distribution patterns of these species, while outlining habitat units that may be important for population recovery. At its simplest call presence represents whale presence. Additionally, recordings show a strong temporal separation in call type employed by fin whales. Their stereotypical 20 Hz call dominates recordings from December to late February, with presence of patterned sequences representing a doublet song also. Recordings made later in the spring (March-April) show high prevalence of the 40 Hz call, described for social and foraging behaviours. These calls were most frequently heard in recordings from along the continental shelf break and in areas of topographical complexity and possible prey aggregation, such as oceanic canyons. Change in call type suggests a change in behaviour and social context of the signaler. Similar patterns were found for blue whales in this area; more B calls in winter and D calls in spring.

11:45

5aAB12. Characterizing the acoustic behavior of free-ranging Risso's dolphins (*Grampus griseus*) in Monterey Bay, California. Brijonnay C. Madrigal and Alison K. Stimpert (Vertebrate Ecology Lab, Moss Landing Marine Labs., 8272 Moss Landing Rd., Moss Landing, CA 95039, bmadrigal@mlml.calstate.edu)

Risso's dolphins (*Grampus griseus*) are a common, highly vocal odontocete species found in Monterey Bay, California, that is relatively understudied acoustically. Although several studies have focused on Risso's echolocation, there is little research on the other social sound types for this species. Apart from echolocation clicks, the two most common sound types in Risso's dolphin repertoires are whistles and whistle + burst pulse (whistle BP) vocalizations. In summer 2017, single C57 omnidirectional hydrophone (Cetacean Research Technology) deployments were conducted in Monterey Bay to record Risso's sound production during periods of slow travel and social interaction at the surface. Group composition information and surface behavioral events were also recorded to provide behavioral context. Average group sizes consisted of approximately 30 animals. We completed a total of 62 h on effort and 75 deployments. Of the 5 total hours of recordings, 43% contained vocalizations. We will describe the acoustic parameters of recorded whistle and whistle BP vocalizations of Monterey Bay Risso's dolphins, evaluate sound production rate in relationship to surface active behavior, and compare vocalizations with those of geographically isolated populations as well as sympatric odontocete species in the Bay.

Session 5aAOa

Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Ocean Observatories: Laboratories for Acoustical Oceanography I

Thomas Dakin, Cochair
Ocean Networks Canada, TEF-128A 2300 McKenzie Ave., University of Victoria, Victoria, BC V8W2Y2, Canada

Kay L. Gemba, Cochair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, CA 92037

Chair's Introduction—7:45

Contributed Papers

7:50

5aAOa1. An overview of non-governmental organization, passive acoustic, ocean observatory data available on the North East Pacific Coast. Thomas Dakin (Ocean Networks Canada, TEF-128A 2300 McKenzie Ave., University of Victoria, Victoria, BC V8W2Y2, Canada, tdakin@uvic.ca), Janie Wray (Fin Island Marine Inst., Hartley Bay, BC, Canada), Hermann Meuter (CetaceaLab, Hartley Bay, BC, Canada), Jordan Wilson (Pacific Wild, Bella Bella, BC, Canada), Larry Peck (Saturna Island Marine Res. and Education Society, Saturna Island, BC, Canada), Scott Veirs (Orcasound, Seattle, WA), and John P. Ryan (MBARI, Moss Landing, CA)

There are governmental (DFO, DND, NOAA, USN), commercial (typically proprietary), and non-governmental organization (NGO) passive acoustic ocean observing systems installed on the North East Pacific Coast. The NGO passive acoustic datasets span many years, four decades for OrcaLab. This presentation provides an overview of the data collected at various NGO acoustic observatories on the North East Pacific Coast. The dataset time windows, sampling rates, formats, and ancillary data are given for each NGO. Data availability, searchability, data use restrictions, and contact information are given to allow acoustic researchers to access this treasure of acoustic observatory data. NGO's include Fin Island Marine Institute, CetaceaLab, Pacific Wild, OrcaLab, Ocean Networks Canada, Saturna Island Marine Research and Education Society, Orcasound, and the Monterey Bay Aquarium Research Institute.

8:05

5aAOa2. Passive ocean sensing and soundscape analysis of the Juan de Fuca plate using the Ocean Networks Canada NEPTUNE Array. Christopher M. Verlinden (OASIS Inc., 5 Militia Dr., Clarksburg, MA 02421, cmaverlinden@gmail.com), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA), Martha C. Schonau (OASIS Inc., Falls Church, VA), and Thomas Dakin (Univ. of Victoria, Victoria, BC, Canada)

The Ocean Networks Canada (OCN) deployment of a small set of acoustic recorders (4) in and around the Juan de Fuca plate provides the opportunity to observe the seasonal and long-term trends in the northeast Pacific ocean soundscape. With one receiver near the opening of the Straits of Juan De Fuca, two on the continental slope and one in deep water, we have the opportunity to evaluate the effects of local environment on the ambient sound. Wind waves and surface ships drive the low frequency ambient sound fields in most regions. These long-term recordings provide the opportunity to perform passive acoustic tomography, where the cross-correlation of the acoustic field over long time periods can provide estimates of the channel impulse response between the two hydrophones. This technique has been demonstrated experimentally at shorter ranges. The expected oceanographic observed signal from a "tomography experiment" is estimated using the ECCO4 model and the parabolic equation.

Invited Papers

8:20

5aAOa3. Overview of OOI/RSN cabled ocean observatory. Gerald Denny, Michael Harrington, Dana Manalang (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Henderson Hall, Seattle, WA 98105-6698, denny@apl.washington.edu), and Deborah Kelley (School of Oceanogr., Univ. of Washington, Seattle, WA)

The summer of 2014 saw the completion of installation and first data from the cabled portion of the Ocean Observatories Initiative (OOI) Regional Scaled Nodes (RSN). Data streams live from over 130 instruments to an OOI Data Portal via telecom cables that land in Pacific City, OR. The observatory encompasses a cross-section of the Juan de Fuca plate from the coastal shelf to an undersea volcano, with instruments from 80 m to 2900 m depth. Acoustic instrumentation includes 6 broadband hydrophones (capable of 120 kHz), 5 low frequency hydrophones (to 5 kHz), two 3-frequency upward looking fisheries sonars, and 6 upward looking ADCPs. Water-borne acoustic data have also been observed in the 5 broadband (3 mHz to 10 Hz) and 7 short-period (0.1 Hz to 100 Hz) seismometers. Two new acoustic experiments will be deployed in the 2018 O&M cruise. The raw and processed data are archived and are open access to all. An overview of the types of instruments and their locations, as well as discussion of data characteristics and how to access the data, will be given. Some examples of data will be shown as well as examples of specific event characterization and ongoing uses.

8:40

5aAOa4. Low-frequency acoustic observations with cabled observatories on the Juan de Fuca Ridge. William S. Wilcock (School of Oceanogr., Univ. of Washington, Box 357940, Seattle, WA 98195, wilcock@uw.edu), Maya Tolstoy (Lamont-Doherty Earth Observatory, Columbia Univ., Palisades, NY), Robert Dziak (Pacific Marine Environ. Lab., National Oceanic and Atmospheric Administration, Newport, OR), Del Bohnenstiehl (Dept. of Marine, Earth, and Atmospheric Sci., North Carolina State Univ., Raleigh, NC), Jacqueline Caplan-Auerbach (Geology Dept., Western Washington Univ., Bellingham, WA), Felix Waldhauser (Lamont-Doherty Earth Observatory, Columbia Univ., Palisades, NY), and Christian Baillard (School of Oceanogr., Univ. of Washington, Seattle, WA)

In the Northeast Pacific Ocean, the NSF Ocean Observatories Initiative (OOI) and Ocean Networks Canada (ONC) operate regional cabled observatories that include a network of seismometers and hydrophones enclosing the northern half of the Juan de Fuca plate, and local observatories at two sites on the Juan de Fuca Ridge: Axial Seamount (OOI) and the Endeavour segment (ONC). At each ridge site, local networks of seismometers and hydrophones provide a tool to monitor in real time the seismic and acoustic signals associated with volcanic, tectonic, and hydrothermal processes. In 2015, Axial Seamount erupted a few months after the installation of the OOI. Following the eruption, regionally recorded T-phases were used to track a dike propagating along the Axial North Rift. Acoustic signals from impulsive sources associated with lava flows, detected eruptions in the caldera and on the North Rift. Diffuse acoustic signals were also associated with the later stages of the eruption. The low-frequency acoustic capabilities of the OOI and ONC cabled observatories are underutilized and future applications could include the reestablishment of regional acoustic earthquake monitoring, tracking eruptions in real time to guide rapid autonomous and ship-based response efforts, and enhanced studies of fin and blue whales.

Contributed Papers

9:00

5aAOa5. Adding value to big acoustic data from ocean observatories: Metadata, online processing, and a computing sandbox. Ben Biffard, Michael Morley, Maia Hoeberechts, Allan Rempel, Thomas Dakin, Richard K. Dewey, Reyna Jenkyns (Ocean Networks Canada, PO BOX 1700 STN CSC, Victoria, BC V8P 1V1, Canada, bbiffard@oceannetworks.ca),

Ocean Networks Canada (ONC) operates ocean observatories on all three of Canada's coasts. The instruments produce 300 gigabytes of data per day with over 600 terabytes archived so far. The majority of this data is acoustic, both passive (335 TB) and active (20 TB). This demonstrates the unprecedented capability of cabled observatories to provide unlimited power and data for high bandwidth, continuous data acquisition. Handling this data is a challenge. Metadata, calibration, quality control, and access must be considered. The volume of data is too great for most users to handle. Even if they could store and process it, data transfer to users' computers is a limiting, and perhaps unnecessary step. To address these challenges, ONC has developed a data portal, known as Oceans 2.0, that includes on-demand user-configurable online previewing and processing and a computing "sandbox" where users can upload their own code to process the data. The data portal is now fully accessible by web services. The sandbox is a contained, secure environment with direct access to the data. This paper will present our experience and best practices, including use cases, from acquisition to adding value to the data with these new computing methods.

9:15

5aAOa6. Low-frequency ambient noise trends of (almost) 2 decades in the northern Pacific Ocean. Rex K. Andrew (Appl. Phys. Lab., 1013 NE40th St., Seattle, WA 98105, rx.andrew@ieee.org), Bruce Howe (Dept. Ocean Res. Eng., Univ. of Hawai - Manoa, Honolulu, HI), and James Mercer (Appl. Phys. Lab., Seattle, WA)

Nearly two decades of low-frequency (20–500 Hz) ambient noise measurements at seven open-ocean sites in the North Pacific Ocean basin have revealed a complex pattern of long-term trends. The trends in the Northeastern Pacific Ocean show a significant decrease of almost 2 dB/decade. Along the Aleutian archipelago, the levels are either slightly increasing or remaining flat. Levels in two north central Pacific Ocean sites are essentially flat. Comparisons with very sparse measurements made over the last 5 decades suggest that the mid-latitude noise levels may have peaked in the 1990s. These measurements also show, however, that the noise level is still rising elsewhere. The mechanisms driving these trends appear to be more subtle than simply the number of merchant ships or the local wind speed. Climatically-influenced basin-scale acoustic propagation conditions may have an important role.

9:30

5aAOa7. Long-term monitoring of marine soundscapes: Shipping, biodiversity, and weather at a Pacific seafloor observatory. Alice Richards (Phys., Univ. of Bath, Bath, Bath and NE Somerset, United Kingdom) and Philippe Blondel (Phys., Univ. of Bath, Claverton Down, Bath, Avon and NE Somerset BA2 7AY, United Kingdom, p.blondel@bath.ac.uk)

Seafloor observatories enable long-term monitoring of marine soundscapes, modulated by weather, biodiversity, and human impacts (e.g., shipping). High-frequency (96-kHz) measurements at the NEPTUNE node of Folger Deep are processed to compare signatures of shipping and natural events over five tidal cycles, spanning several seasons (2009–2011). Independent meteorological data from local surface stations is also used. Sound levels in third-octave frequency bands centred on 63 Hz and 125 Hz are used to monitor shipping, in line with the European Marine Strategy Framework Directive (MSFD). The contribution from the busy shipping lane 40 km away is affected by the complex, shallow bathymetry, whereas local traffic can increase noise levels by up to 30 dB re. 1 μPa, with strong seasonal variation. The relative contributions of the 63 Hz and 125 Hz bands varied contrary to MSFD expectations for deeper areas. Our results match other studies in shallow, coastal environments, showing the importance of depth in interpreting changes. Principal-Component Analyses show that noise from local vessels is the most significant contributor in all seasons, and weather is the second largest, except in summer when biological noise became prevalent. Biodiversity, measured with the broadband Acoustic Complexity Index, showed a strong correlation with weather.

9:45–10:00 Break

10:00

5aAOa8. Ranking vessel noise emissions using measurements from an underwater listening station. David E. Hannay, Heloise Frouin-Mouy, Zizheng Li, and Alexander O. MacGillivray (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, David.Hannay@jasco.com)

Commercial shipping routes pass through important habitat areas for several species of marine mammals in the coastal waterways of southern British Columbia. The Vancouver Fraser Port Authority, through its Enhancing Cetacean Habitat and Observation (ECHO) program, has undertaken studies to develop mitigation measures that will lead to a quantifiable reduction in threats to whales resulting from shipping activities. This includes long-term measurements of vessel noise at a cabled underwater

listening station in Georgia Strait (the ECHO ULS) where JASCO, in partnership with Ocean Networks Canada, has been measuring source levels of vessels calling at the Port using JASCO's PortListen® software. PortListen® receives and processes real-time acoustic and AIS data to calculate vessel source levels using ANSI standard methods (S12.64-2009 R2014). Since September 2015, PortListen® has collected a database of thousands of source level measurements, which has been used to implement ranking system for vessel noise emissions. The ranking system uses a data-driven model to adjust the ranking of each measurement according to the vessel characteristics (e.g., size, class) and measurement conditions (e.g., speed, wind, and draft). In addition to an unweighted noise ranking, the system also provides weighted rankings for five marine mammals hearing groups using NOAA (2016) auditory weighting curves.

Invited Paper

10:15

5aAOa9. The Aloha Cabled Observatory: New Insights into hurricane generation of the seismo-acoustic noise spectrum. Rhett Butler (SOEST, Univ. of Hawaii at Manoa, 1680 East-West Rd., POST 602, Honolulu, HI 96822, rgb@hawaii.edu) and Jérome Aucan (Laboratoire d'Etudes en Géophysique et Océanographie Spatiale, Institut de Recherche pour le Développemen, Toulouse, France)

The close passage of Hurricane Lester near the Hawaiian Islands in September 2016 afforded an in-depth, close-up study of storm generation of the largest background vibrations observed planet wide. The observations at the ALOHA Cabled Observatory on the seafloor below the Hurricane, coupled with seismic sensors on Oahu, and ocean wave buoys off shore, present a detailed picture connecting the storm to the ocean and Earth. Wave interactions from a distant typhoon near Japan play an important role. Vibration energy levels observed on Oahu closely match those on the sea floor 100 km north of Oahu, where ALOHA Cabled Observatory is the world's deepest seafloor observatory at 4,728 m depth. Characteristic vibrations generated radially from the Hurricane were observed, along with unexpected transverse motions perpendicular to the radial waves. This latter observation is consistent with a broad source region extending from Hurricane Lester and generating the vibrations. Evidence for substantial scattering of the vibrations in the ocean crust is inferred, due to slanting layers and directionally varying velocities, dating back nearly 80 million years ago when the sea floor was being originally being emplaced at a Pacific mid-ocean ridge. This hurricane transit yields new knowledge on how storms vibrate the planet.

Contributed Papers

10:35

5aAOa10. Long-term marine mammal occurrence in the Distributed Biological Observatory 2010–2015. Catherine L. Berchok (Marine Mammal Lab, Alaska Fisheries Sci. Center/NOAA, 7600 SandPoint Way NE, Seattle, WA 98115, cberchok@yahoo.com), Eric K. Braen (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, Seattle, WA), Jessica Crance (Marine Mammal Lab, Alaska Fisheries Sci. Center/NOAA, Seattle, WA), Stephanie L. Grassia, Jenna M. Harlacher, Eliza G. Ives, Jessica M. Kimber, Julie A. Mocklin, Megan A. Wood, Dana Wright (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, Seattle, WA), Phyllis J. Stabeno (NOAA, Pacific Marine Environ. Lab., Seattle, WA), and Sue E. Moore (NOAA/S&T-PMEL, Seattle, WA)

The Distributed Biological Observatory (DBO) is a set of eight biological hotspot areas spanning latitudinally from the Northern Bering Sea to the Canadian Beaufort Sea. The DBO is an international collaboration between researchers from the United States, Japan, Canada, China, South Korea, and Russia that work in the Alaskan Bering, Chukchi, and Beaufort Seas; all research vessels passing through one of the DBO regions collect biophysical data (i.e., temperature, salinity, sea ice concentration and thickness, chlorophyll, nutrients, and zooplankton occurrence) along a pre-described line of sampling stations. Since the pilot study in 2010, and with funding from the Bureau of Ocean Energy Management (BOEM), the Marine Mammal Laboratory at the Alaska Fisheries Science Center of NOAA has maintained passive acoustic recorder moorings at two of the DBO regions, colocated with oceanographic moorings from the Pacific Marine Science Center (P. Stabeno). This coverage was expanded to five DBO regions in 2012, again with colocated oceanographic moorings at four of the sites. Here, an interannual comparison of the long-term mooring results from gray, bowhead, beluga, humpback, and killer whales, walrus, ribbon and bearded seals, and vessel and seismic airgun noise will be presented and compared with the sampled biophysical data.

10:50

5aAOa11. Orcasound lab: A soundscape analysis case study in killer whale habitat with implications for coastal ocean observatories. Scott Veirs (NEMES, Univ. of Victoria, 7044 17th Ave. NE, Seattle, Washington 98115, sveirs@gmail.com), Val Veirs (Beam Reach (SPC), Friday Harbor, WA), Lauren McWhinnie, Patrick O'Hara, and Gregory O'Hagan (NEMES, Univ. of Victoria, Victoria, BC, Canada)

Orcasound lab is a cabled hydrophone array located near the shoreline of Haro Strait, the core summertime habitat of the endangered southern resident killer whales (SRKWs). In 2016–2017, we began to record data continuously on local hard drives and in 2018 are archiving both lossy and lossless data 24/7 in an AWS/S3 bucket. We discuss our statistical characterization of the soundscape from these continuous audio recordings, contextualized with the AIS data (to quantify sources of ship noise) and image data (to quantify sources of non-AIS boat noise). Of particular interest to ocean observatories are our methods of establishing non-anthropogenic acoustic baselines and then ranking noise pollution sources relative to these baselines. We explore the statistical consequences of selecting different averaging times (from seconds to years) and frequency band widths (spectrum to broadband levels) when computing baselines and pollution metrics, including "delta" metrics that may be most-relevant to SRKWs. Finally, we explain how soundscape analysis (with attention to tidal, diurnal, seasonal, or decadal time variations) could be implemented with cloud-based data in near-real-time and be enriched by citizen scientists interacting with a time-stamped live audio stream and other environmental data.

Invited Papers

11:05

5aAOa12. Joint observatories following a single male Cachalot during 12 weeks —The Yukusam story. Paul Spong, Helena Symonds (OrcaLab, Alert Bay, BC, Canada, orcalab2@gmail.com), Herve Glotin (LIS, CNRS, AMU, Univ Toulon, La Garde, France), Jared Towers (Fisheries & Oceans Canada, Alert Bay, BC, Canada), lisa larsson (OrcaLab, OrcaLab, BC, Canada), Thomas Dakin (Univ. of Victoria, Victoria, BC, Canada), Scott Veirs (Orcasound.net, Seattle, Washington), Elizabeth Zwamborn (Dalhousie, Halifax, NS, Canada), james pilkinton (Fisheries & Oceans Canada, Victoria, BC, Canada), Pascale Giraudet (LIS, CNRS, AMU, Univ Toulon, Toulon, France), Val Veirs (Orcasound.net, Friday Harbor, Washington), Jason Wood (SMRU, Friday Harbor, Washington), and John Ford (Fisheries & Oceans Canada, Victoria, BC, Canada)

From 11 February to 31 March 2018, a lone male sperm whale visited coastal waters from the northeast to southern ends of Vancouver Island. This whale, named "Yukusam" after the Namgis First Nation word for Hanson Island, near where the whale was first observed and recorded, is the first sperm whale recorded acoustically in the area since 1984 and is the only sperm whale ever observed in coastal waters between Vancouver Island and continental North America. The Yukusam tracking story is a showcase for the potential of acoustic observatory collaborations. Tracking a single animal over such time and distance is remarkable. It obviously helped that Yukusam was the only sperm whale in the area, but still, the experience hints at the potential for using diverse independent observatories collaboratively. We then aim to see all the observatories running automated detection classification and location software and having all the data tied in to a public database for a total of almost 500 Gb of recordings, with labels. Supplemental material @ http://sabiod.org/yukusam

11:25

5aAOa13. Single-hydrophone automated passive acoustic ranging of fin whales at Station ALOHA. Brendan P. Rideout (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bprideou@hawaii.edu) and Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

This paper presents a technique for performing passive underwater acoustic ranging with data from a single hydrophone and builds upon earlier localization approaches which estimate the sound source position using times of arrival of acoustic energy traveling along direct and/or interface-reflecting paths between source and receiver. In this work, measured time differences between interface-reflecting and direct path arrival times are compared with a set of model-predicted time differences calculated over a set of candidate source ranges in a way that does not require measured arrival paths to be labeled (e.g., direct, surface bounce, bottom bounce, etc.). The modeled set with the best match to the measured data indicates the best estimate of source range. To enable the processing of multi-year data sets, the detection and localization steps are automated and, where possible, multi-threaded to improve computational efficiency on multi-core computer processors. This approach is demonstrated using 20-Hz fin whale (Balaenoptera physalus) calls recorded by the ALOHA Cabled Observatory (ACO), 100 km N of Oahu (Hawaii) in 4782 m of water.

FRIDAY MORNING, 9 NOVEMBER 2018 ESQUIMALT (VCC), 8:00 A.M. TO 10:45 A.M.

Session 5aAOb

Acoustical Oceanography and Underwater Acoustics: Experimental Assessment of Theories of Sound Propagation in Sediments I

Orest Diachok, Cochair
Johns Hopkins University APL, 11100 Johns Hopkins Rd., Laurel, MD 20723

N. Ross Chapman, Cochair
School Earth and Ocean Sciences, Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada

Chair's Introduction—8:00

Invited Papers

8:05

5aAOb1. Model/data comparisons and numerical experiments for the comparison of sandy sediment models. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.l.bonomo@navy.mil) and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Many geoacoustic models have been proposed to study the acoustic behavior of sandy sediments to predict the propagation within the sediment and the reflection loss from waterborne waves. Sandy sediments have been described as a two-phase porous medium using the theoretical framework developed by Biot. Other proposed theories were developed under the assumption that sandy sediments

possess no effective skeletal matrix and instead describe the acoustic behavior of the sediments using grain-to-grain contacts. This talk summarizes model/data comparisons and numerical experiments in an attempt to determine which of these two theoretical frameworks is more suitable for the modeling of sandy sediments. Recommendations are made for future experiments that can aid in resolving this issue. [Work supported by ONR, Ocean Acoustics.]

8:25

5aAOb2. Seabed reflection measurement results shed light on theories of acoustic propagation in marine sediments. Charles W. Holland (Appl. Res. Lab., The Pennsylvania State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer (Univ. of Calgary, Calgary, AB, Canada), and Stan E. Dosso (Univ. of Victoria, Victoria, BC, Canada)

Marine sediments are inherently a complex assemblage of solid particles (of size ranging over five orders of magnitude) with fluid-filled interstices sometimes containing gas. The mechanisms that govern the dispersive wave speeds and concomitant attenuation are still hotly debated. In order to shed light on the mechanisms and current theoretical approaches to approximating them, measurements of in-situ sediments are desirable since they provide the natural complexity desirable for testing theories against realistic sediment structures. On the other hand, in-situmeasurements present significant challenges since seabed effects must be separated from other ocean processes, e.g., water column variability, biologics and sea surface roughness, and bubbles. Seabed reflection measurements offer one way to mitigate unwanted ocean processes. Reflection-derived observations are presented that shed light on current sediment acoustic models. In addition, experiments are proposed which will further test and help guide sediment acoustics theoretical developments. [Work supported by the ONR Ocean Acoustics Program.]

8:45

5aAOb3. Modeling granular sediments—Experimental results that support the physics included in "Biot-like" models, other possible experiments that could be interesting. Kevin Williams (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Under the "all models are wrong but some are useful" philosophy, one may at times allow some abuse of the basic physics if the predictions are accurate. However, few would start with a flawed model when they need not. We start with a thought experiment that indicates a fundamental requirement for capturing the correct dynamics in modeling porous granular media and show that this physics is captured in Biot and Pierce/Carey models as well as models that use them as a starting point. Ocean and laboratory propagation experiments that test the predictions of those models are then described. Since other modeling choices can, given free parameters, predict both granular sound speed and attenuation, we continue by looking at both back and forward scattering results. These results further mandate the need for a model that correctly treats the sediment dynamics. Finally, we present possible additional laboratory experiments that could alter some of the most important parameters inherent in Biot-like models and thus further test the validity of the such models as starting points in granular sediment acoustics. [Work supported by the Office of Naval Research.]

9:05

5aAOb4. Laboratory and at-sea experiments using a vertical synthetic array technique to measure the sound waves propagating above and below fluid-saturated sediment interfaces. Harry J. Simpson and Brian H. Houston (Physical Acoust. Branch, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, harry.simpson@nrl.navy.mil)

A series of measurements in the NRL shallow water laboratory were designed and conducted to measure the sound waves propagating in an un-consolidated fluid-saturated porous medium. The novel design used a pair of small receivers, one buried in the bottom and one mounted in the water column to a linear vertical positioner. The buried receiver was carefully inserted into the sandy bottom using a small water-jet, engineered to have no entrained air, to minimally disturb the bottom. The receiver pair were incrementally pulled up through the bottom or water column to map out the sound propagating from a source mounted in the water column. The position of the source was varied to measure above critical angle and below critical angle sound penetration into the bottom. Measurements of carefully smoothed and roughened interfaces were also conducted. This synthetic vertical array technique proved very successful in the laboratory and was transitioned to at-sea measurements in a variety of sandy bottom locations around Panama City, FL. The analysis and modeling of these laboratory and at-sea experiments will be discussed.

9:25

5aAOb5. An overview of the sediment acoustic models and their experimental validations. Altan Turgut (Naval Res. Lab, Acoust. Div., Code 7160, Washington, DC 20375, altan.turgut@nrl.navy.mil)

The Biot theory of acoustic wave propagation in porous media was first adapted to marine sediments by Stoll (1969). To describe the response of a slightly inelastic skeletal frame over a wide frequency range, Stoll introduced the notion of constant complex bulk and shear moduli with small imaginary parts. Stoll's model predicts f^1 frequency-dependence of attenuation at low frequencies and nearly f^2 and $f^{1/2}$ frequency-dependence at frequencies where maximum velocity dispersion occurs. Although the theory well reproduces the velocity and attenuation measurements in marine sediments, the assumption of constant complex moduli slightly violates the causality (Turgut, 1990). The more recent Grain-Shearing (GS) and Viscous Grain-Shearing (VGS) models (Buckingham, 2000 and 2007) are causal and the VGS model also predicts a frequency-dependence of attenuation like that of the Stoll model. With the selection of proper parameter values, both models predict compressional and shear-wave dispersions that are in agreement with those of previous *in-situ* and laboratory measurements. In addition, the Stoll model predicts the existence of slow compressional waves that have been observed in synthetic porous media (Plona, 1980) but not in natural marine sediments. Several reflection and in-sediment transmission experiments are discussed to facilitate the detection of slow compressional waves in marine sediments. [Work supported by ONR.]

9:45–10:00 Break

10:00

5aAOb6. Uncertainty quantification and spatial variability of velocity- and attenuation-frequency dependence along a 14-km seabed survey on the Malta Plateau. Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, jan.dettmer@ucalgary.ca), Charles W. Holland (Appl. Res. Lab., The Pennsylvania State Univ., State College, PA), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We study compressional-wave frequency dependence of sound velocity and attenuation in the seabed by inverting reflectivity data recorded by an autonomous underwater vehicle (AUV) on the Malta Plateau along a 14-km survey track. The AUV towed a 32-hydrophone array and a source emitting signals at ~4-m intervals in two frequency bands (900–1300 and 1900–3600 Hz). The reflection data are processed in terms of reflection coefficients which results in ~1500 data sets, each with a seabed footprint of $<$20 m. For efficient Bayesian uncertainty quantification, a trans-dimensional particle filter is applied. The dataset provides a usable frequency bandwidth of 1000–3400 Hz to study velocity- and attenuation-frequency dependence which is modelled with viscous grain shearing theory. The trans-dimensional model allows frequency-dependence inferences as a function of depth while fully accounting for the unknown seabed stratification which substantially affects the estimates. Finally, the AUV acquisition provides the means to study the frequency-dependent seabed variability at mesoscales of several meters which are poorly understood. [Data are from CLUTTER JRP, a collaboration of ARL-PSU, DRDC, CMRE, and NRL. Research supported by the Natural Sciences and Engineering Research Council of Canada.]

10:15

5aAOb7. *In situ* measurements of compressional and shear wave propagation in marine sediments and comparison to geoacoustic models. 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), Gabriel R. Venegas (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

In situ measurements of compressional and shear wave speed and attenuation provide direct characterization of marine sediment acoustic properties at ambient conditions, as opposed to measurements that are conducted on samples removed from the seabed. Sediment geoacoustic models, such as those based on fluid, viscoelastic, poroelastic, or suspension wave theories, relate physical properties of the seabed like porosity, grain size, and pore fluid viscosity to the wave speeds and attenuation. Other models can include the effects of gas bubble distributions, which can be associated with the presence of benthic biology or organic matter decomposition, on compressional and shear wave propagation. In this paper, we examine sediment geoacoustic models in comparison with *in situ* acoustic data from various field experiments, for sediments containing various ratios of sand, silt, clay, and organic matter. The model input parameters are partially constrained by geotechnical data obtained from core samples collected at the field experiment sites. The applicability of various models to the sediments encountered at each measurement site will be discussed. [Work supported by ONR and ARL:UT IR&D.]

10:30

5aAOb8. Speed of sound and geo-alpha in clayey silt derived from concurrent inversion of geo and bio-acoustic parameters from transmission loss data and co-located chirp sonar measurements. Orest Diachok (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com) and Altan Turgut (Naval Res. Lab., Washington, DC)

Application of the concurrent geo and bio-acoustic inversion method (Diachok and Wales, 2005) to broadband (0.3–5 kHz) transmission loss (TL) measurements, and co-located normal incidence chirp sonar measurements provided measures of geo-acoustic properties of clayey-silt in the Santa Barbara Channel. Inversion calculations assumed that the geological environment may be characterized by the interfacial sound speed, unconsolidated layer thickness, sound speed gradient, g, and geo-alpha, α_G; and that the biological environment may be characterized by the layer depth, layer thickness, and bio-alpha (attenuation coefficient within the layer). Co-located cores provided *ground truth*; echo sounder and trawl measurements provided *biological truth*. Both the concurrent inversion and the chirp-based methods yielded approximately the same value of g, 6.5/s, in the top 9 m thick layer of clayey-silt. The value of α_G, based on TL measurements, was quite low, approximately 0.02 dB/λ, in good agreement with Holland and Dosso's (2013) only previously reported *in-situ* estimate of geo-alpha in silty-clay in this frequency-depth range. These results have important implications for estimation of geo parameters from TL measurements in biologically intense environments. Refinements to envisioned follow-on experiments will be discussed. [This research was supported by the Office of Naval Research Ocean Acoustics Program.]

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics I

Stephanie Konarski, Chair
Naval Research Laboratory, Washington, DC

Contributed Papers

9:00

5aPA1. Effective dynamic properties of random complex media. M. Mahbub Alam, Francine Luppé (Laboratoire Ondes et Milieux Complexes, Univ. of Le Havre, Le Havre, Normandie, France), Valerie J. Pinfield (Chemical Eng. Dept., Loughborough Univ., UK, Loughborough University, Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk), and Pierre Maréchal (Laboratoire Ondes et Milieux Complexes, Univ. of Le Havre, Le Havre, Normandie, France)

In recent years, it has been demonstrated that, as particle concentration in a suspension increases, the viscous nature of the host fluid becomes more and more significant and thereby needs to be taken into consideration when calculating effective properties for acoustic propagation. One approach to the problem is to incorporate wave conversion phenomena, primarily between compressional and shear wave modes, into the models for effective properties. A self-consistent method that includes mode conversion, based on a core-shell model, has been used to determine the effective dynamic bulk modulus and mass density of random complex media consisting of spherical inclusions. The analytical expressions obtained are compared with those obtained from numerical solution of the self-consistent model equations. The contribution of mode conversion on the effective dynamic properties for particles in a viscous liquid are explored. The variation of the effective properties with frequency, volume concentration, and density contrast are also investigated numerically.

9:15

5aPA2. Experimental measurement of tortuosity, viscous, and thermal characteristic lengths of rigid porous material via ultrasonic transmitted waves. Mustapha Sadouki (Departement des Sci. de la Matière, Université Djilali Bounaama à khemis-miliana, Université Djilali Bounaama à Khemis-Miliana, Rte. Thenia el Had, Ain Defla, Khemis-miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz)

An inverse method is proposed for measuring tortuosity, viscous, and thermal characteristic lengths of air-saturated porous material with rigid frame via ultrasonic transmitted waves at normal incidence. The equivalent fluid model is considered. The interaction between the fluid saturated the pores and the structure are taken into account in two frequency response factors: the dynamic tortuosity of the medium introduced by Johnson et *al*. and the dynamic compressibility of the air introduced by Allard. Simplified expression of the transmission coefficient is obtained in frequency domain, and this expression depends on the porosity, tortuosity, viscous, and thermal characteristic lengths. The inverse problem is solved numerically in time domain by minimizing between simulated and experimental transmitted waves. The inverted parameters are in good agreement with those obtained using conventional methods. Simulated signals are reconstructed using the optimized values found and compared with the experimental signals. Tests are performed using three different plastic foam samples having low flow resistivity. The proposed technique has the advantage of being simple, fast, and not expensive.

9:30

5aPA3. Fractional viscoelastic modelling of wave propagation in fluid-saturated marine sediments. Vikash Pandey (Ctr. for Ecological and Evolutionary Synthesis (CEES), Dept. of BioSci., Univ. of Oslo, Postboks 1080, Blindern, Oslo 0316, Norway, vikashp@ifi.uio.no)

It is shown that Buckingham's grain-shearing (GS) model [JASA (2000)] as well as its improved viscous-GS (VGS) model [JASA (2007)] can be expressed using the mathematical framework of fractional calculus. The fractional version of the standard fluid model is adopted to independently arrive at the wave equations and dispersion relations derived by Buckingham. The fractional-order wave equations obtained for the compressional waves and shear waves are relatively easier to analyse due to their closed-form representation in the fractional framework. It is also shown that the fractional calculus approach may help in bridging the disparate fields of non-Newtonian rheology and sediment acoustics, which may have actually developed independently of each other. Further, the experimental data relating wave dispersion and attenuation in marine sediments is found to match with the predictions from the fractional framework. The overall goal is to show that fractional calculus is not just a mathematical framework that can only be applied to curve-fit the observational data for complex media. In fact, it has an inherent connection to real physical processes that needs to be explored more.

9:45

5aPA4. Control and evaluation of liquid crystal molecules using ultrasound vibration. Yuki Harada (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, ctuc1005@mail4.doshisha.ac.jp), Daisuke Koyama, Hirokazu Yasui, Marina Fukui, Yuki Shimizu, Yoshiaki Shibagaki, and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

Nematic liquid crystals are widely used in optical devices such as liquid crystal displays and controlled by electric fields through the liquid crystal layer. The authors have proposed a technique to control the orientation of liquid crystal molecules using ultrasound vibration without indium tin oxide electrodes. An ultrasound liquid crystal cell was fabricated; it consists of a liquid crystal layer sandwiched by two glass plates having two ultrasound PZT transducers. When exciting the transducers at the resonance frequencies, the flexural vibration modes were generated on the cell and the acoustic radiation force acted to the liquid crystal molecules so that the orientation direction can be changed. The relationship between the molecular orientation and the vibration distribution of the cell was investigated from the transmitted light distribution of the liquid crystal cell under the crossed Nicol condition. In addition, the orientation of liquid crystal molecules was evaluated as ultrasonic wave velocity in the GHz range by the Brillouin scattering method. The ultrasonic wave velocity in the long axis direction of the liquid crystal molecules was higher than that in the short axis direction, which indicates the uniaxial anisotropy of the liquid crystal molecules.

10:00

5aPA5. Ultrasound manipulation of 3D dynamic user-specified patterns of particles in air. Milo Prisbrey and Bart Raeymaekers (Mech. Eng., Univ. of Utah, 201 Presidents Circle, Salt Lake City, UT 84119, mprisim@gmail.com)

Ultrasound noncontact particle manipulation (NPM) employs the convergence of the acoustic radiation force associated with an ultrasound wave field to levitate and manipulate particles in a fluid medium. We use multiple phased arrays of ultrasound transducers to dynamically manipulate a 3D pattern of particles along a user-specified trajectory. We numerically simulate the ultrasound NPM method and experimentally validate it by creating dynamic 3D patterns of expanded polystyrene particles in air, and observe good quantitative agreement. The method allows manipulating an entire user-specified pattern of particles and sub-sets of a pattern to traverse a user-specified trajectory in 3D. This experimental demonstration shows that ultrasound NPM can be implemented in engineering applications such as containerless transport and measurement, and manufacturing of engineered materials.

10:15–10:30 Break

10:30

5aPA6. Photo-thermal transient stress observed by sub-nanosecond pump-probe technique with the surface plasmon resonance. Hiromichi Hayashi, Hayato Ichihashi (Dept. of Elec. and Electronics Eng., Doshisha Univ., 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, ctwb0305@mail4.doshisha.ac.jp), Shoya Ueno (Dept. of Biomedical Information Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Mami Matsukawa (Dept. of Elec. and Electronics Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

Photoacoustic microscopy (PAM) has attracted attention as a non-invasive observation tool for biomedical studies. However, the spatial resolution is unsuitable for precise imaging of a single cell due to the characteristics of the ultrasonic transducer. In recent years, the surface plasmon resonance (SPR) sensor has been reported as a technique with a high-spatial resolution and ultra-flat frequency response. SPR is expected to be the future optical detector for photoacoustic imaging. As a first trial, using a Kretschmann-configuration SPR sensor (Ag (53 nm)/BK7 glass prism), we observed the photo-thermal transient stress and the consequent change in refractive index using the sub-nanosecond pump-probe technique. The signal intensity of the thermo-elastic stress produced by the pump beam excitation was measured by changing the incident angle of the probe beam. The spot diameter of the probe beam was 5 μm. The highest intensity signal was observed at the SPR angle (46.5 deg.) of the plasmon resonance. Results suggest that the SPR sensor effectively enables the non-contact and non-destructive measurements of the thermo-elastic stress. Further researches will implement the SPR sensor for tissue characterization by attaching biological samples on the Ag surface.

10:45

5aPA7. Fast tracking of acoustic resonance frequencies in collection volumes to measure flow. Keith A. Gillis, James W. Schmidt, Jodie G. Pope, and Michael Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov)

Previously, we demonstrated accurate determinations of the mass of gas in a large collection vessel of known volume from measurements of the pressure and the resonance frequencies of a few acoustic modes. We achieved accurate mass measurements even when linear temperature gradients persisted after adding or removing gas. The change in mass Δm from before and after flow and the collection time Δt may be used to calibrate mass flowmeters. Because the resonance quality factors Q are very high (up to 30000), measuring the complex-valued frequency response using a lock-in amplifier to determine the resonance frequency is slow. This static measurement of mass requires sufficient time to elapse for transient non-linear gradients to dissipate through convection and diffusion. Here, we demonstrate a method to dynamically track the resonance frequency of a large, gas-filled collection vessel while gas is flowing into or out of the vessel. We use positive feedback to excite sustained oscillations at the desired resonance frequency without a lock-in amplifier or frequency synthesizer. As the speed of sound changes, the sustained oscillations track the changing resonance frequency f with response time ~$1/f$. Preliminary mass flow measurements using this technique agree with a flow standard within their combined uncertainties.

11:00

5aPA8. Ultrasonic assisted large diameter PBX substitute composites compaction and its mechanical properties. Kezhen Lyu (Inst. of Chemical Mater., No.64, Mianshan Rd.,Mianyang, Sichuan 621999, China, lyukezhen@gmail.com)

The rates of densification and density uniformity are a key performance of PBX parts. Based on ultrasonic vibration technology, this paper designs and manufactures an upper and lower 20 kHz longitudinal and radial vibration assisted molding system with a 200 mm diameter compacting part at the center axis. In this work, PBX substitute composite was successfully fabricated by pressing of powder with ultrasonic vibration and the influence of ultrasonic vibration force coupling on the mechanical properties of PBX substitute powder was investigated. Applying ultrasonic vibration to result in great rates of densification and higher efficiency of the process is expected. The effects of particle size and vibration amplitude and pressure and weight on the densification behavior and of PBX substitute powder are discussed. The results show that applying ultrasonic vibration leads to a higher relative density and an improved homogeneity, and the growth of the size of the pressed sample can be reduced with the ultrasonic vibration and the larger of static pressure, the smaller of the pressed sample size change. In addition, ultrasonic vibration has a great effect on the finer particles and the surface temperature of the sample is not obviously increased with ultrasonic vibration.

11:15

5aPA9. Sound insulation measurement in a tube with broadband pulse secondary transmission method. Mengfei Zhang, Hong Hou, and Nansha Gao (Environ. Eng., Northwestern PolyTech. Univ., 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, yollar@163.com)

Based on time-domain pulse-separation method in a standing wave tube, a method using broadband pulse secondary transmission wave is proposed for sound insulation measurement in this paper. Two kinds of broadband pulse were generated, the incidence wave and *secondary transmission wave* could be obtained easily with one microphone by designing the location of microphone and sample, then acoustic parameters of sample can be calculated. An experiment was completed to verify the proposed method, when compared with time-domain pulse-separation method, a relatively good agreement between these two methods can be observed. Using the proposed method, only *one microphone position and one signal* are required. This method is simple and with good repetition, could be an effective method for sound insulation measurement. Meanwhile, the measurement uncertainty was done.

Session 5aSC

Speech Communication: Development and Clinical Populations (Poster Session)

Meital Avivi-Reich, Chair
Communication Sciences and Disorders, Northwestern University, Room 1-331, Frances Searle Building 2240 Campus Drive, Evanston, IL 60208

All posters will be on display from 8:30 a.m. to 11:30 a.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

Contributed Papers

5aSC1. A biomechanical model for infant speech and aerodigestive movements. Connor Mayer (Linguist, Univ. of California, Los Angeles, 7100 Hillside Ave., Apt. 108, Los Angeles, CA 90046, connormayer@ucla.edu), Ian Stavness (Comput. Sci., Univ. of SK, Saskatoon, SK, Canada), and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

A central question in speech acquisition is how infants are able to learn speech movements rapidly and with limited input. A relatively untested but appealing hypothesis is that some core speech movements may build on pre-existing aerodigestive movements like swallowing and suckling (e.g., MacNeilage, 2008, *The Origin of Speech*; Studdert-Kennedy & Goldstein, 2003, *Launching Language*). We will present a model of an infant tongue and palate using a 3D biomechanical simulation platform (www.artisynth.org; e.g., Stavness *et al.*, 2012, *J. Biomech.* 45(16): 355–394; Gick *et al.*, 2014, *Compu. Meth. in Biomech. & Biomed. Eng.: Imag. & Vis.* 2(4): 217–222). This model, generated from CT and MRI imaging data, will be capable of simulating both swallowing and simple speech movements. The results of simulations using this model will provide useful insight into infant motor control, and will help to supplement neurological, clinical, and kinematic evidence relating speech and aerodigestive movements. [Funding from NSERC.]

5aSC2. An acoustic and perceptual analysis of children's adaptation to an electropalatographic sensor. Shawn L. Nissen, Kasey Duffield, and Marissa Celaya (Commun. Disord., Brigham Young Univ., 128 TLRB, Provo, UT 84602, shawn_nissen@byu.edu)

Previous research has investigated adults' ability to adapt their speech to an electropalatographic (EPG) pseudopalate; however, less is known about how children adapt their speech to the presence of the device. This study examined the acoustic and perceptual effect of an EPG pseudopalate on six elementary school-aged children's ability to produce the fricatives /s/ and /ʃ/. Speech productions were collected at eight time intervals including before placement of the pseudopalate, at 30-min increments with the pseudopalate in place, immediately following removal of the pseudopalate, and 30 min after removal. The fricative targets were produced in a carrier phrase and during spontaneous conversation. Adaptation was evaluated by measuring the fricative duration, spectral mean, spectral variance, and relative intensity, as well as a series of perceptual ratings from 20 native English listeners. Although there was a relatively high amount of variability among and within speakers, for several children evidence of adaptation was found after 30 minutes. For some participants, full adaptation did not occur until the pseudopalate was removed. Although future research is needed, it is hoped that this study will provide a greater understanding of children's ability to adapt to the EPG pseudopalate.

5aSC3. Do children understand adults better or themselves? A perceptual study of Polish /s, ʂ, ɕ/. Marzena Zygis (Leibniz Ctr. - General Linguist, Schützenstr. 18, Berlin D-10117, Germany, zygis@leibniz-zas.de), Marek Jaskula (Westpomeranian Univ. of Technol., Szczecin, Poland), and Laura L. Koenig (Haskins Labs and Adelphi Univ., New Haven, CT)

Developmental studies of speech perception have typically used adult productions or synthetic speech as stimuli. Little is known about how children perceive their own speech, however. Further, few studies of sibilant fricative perception have explored complex sibilant inventories. This work assessed perception of the sibilants /s, ʂ, ɕ/ in 32 monolingual native speakers of Polish, 3–8 years of age. Children participated in a picture-naming task to identify minimal or near-minimal triplets with the fricatives in initial and medial position, e.g., [kasa] "cash point", [kaʂa] "groats", [kaɕa] "Cathe, prop.name." They subsequently labeled their own word productions, and the words as produced by an adult. Children's labeling was generally quite accurate for the adult speaker, with the lowest accuracy rating of ca. 75% seen in the youngest listeners. When labeling themselves, the children were less accurate, with average performance at about 50% in children younger than 55 months. Across all ages, reaction times decreased in the order /ʂ/ > /s/ > ɕ/. Future work will obtain perceptual judgments of the children's productions from other listeners and obtain acoustic measures of the fricatives to explore perception-production relationships.

5aSC4. Phoneme production by Hispanic hearing-impaired children. Tanya Flores (Lang. and Lit., Univ. of Utah, 255 S Central Campus Dr., LNCO 1400, Salt Lake City, UT 84109, Tanya.Flores@utah.edu)

This study examines the speech production of Hispanic deaf and hard of hearing (DHH) children from 4 to 8 years of age who have had delayed medical intervention. The data presented here are from the first year of data collection and will focus on the Spanish and English speech productions of the target group as compared to the productions of the control groups (Hispanic peers with normal hearing and non-Hispanic DHH peers) from the same local community. The goal of the study is to create a speech corpus of Hispanic DHH children that will be used to study various aspects of their language development. Findings will contribute to the currently limited acoustic research on minority DHH children whose home language differs from their specialized language program (in this case, English Listening and Spoken Language Program).

5aSC5. Suprasegmentals affecting fluency levels in elementary students' read-speech: Focusing on pause numbers and pause duration, speech rate, and pitch range. Hyesook Park (Graduate of Education, Yonsei Univ., 1599-1885 50 Yonsei-ro Seodaemun-gu, Seoul 0372, South Korea, pss7078@daum.net)

The purpose of this study was two-fold: 1) to rate the unrated Korean elementary students read-speech (from K-SEC) 2) which and to what extend suprasegmentals—pause numbers and duration, speed rate, and pitch range—would affect the degrees of fluency. To do, 5 raters evaluated 3069 read-speech data into 5 proficiency levels and 500 data were randomly selected for analyzing pause numbers and duration, speed rate, and pitch rage with Praat. The result showed the speech rate is the most influential factor. It means the most fluent readers (5 levels) tend to have the highest speech rate. The next factor is pause duration; the more fluent, the less pause duration one has. Also, statistical regression analysis shows pause numbers and pitch range are not statistically meaningful factors to rate the fluency levels in Korean elementary students read-speech. At the end of this study, some suggestions of the pronunciation drills for Korean elementary students were discussed based on the result of the study.

5aSC6. Global measures of early vocalizations in infants with CP in Taiwan as references to perceptual ratings. Chia-Cheng Lee (Speech and Hearing Sci., Portland State Univ., 1831 SW Park St., Apt 112, Portland, OR 97201, cl9@pdx.edu), Li-mei Chen (Foreign Lang. and Lit., National Cheng Kung Univ., Tainan, Taiwan), Chin-Ting Liu (National Chin-Yi Univ. of Technol., Taichung City, Taiwan), Ray D. Kent (Univ. of Wisconsin–Madison, Madison, WI), Yung-Chieh Lin, Li-Wen Chen, and Ching-Yi Cheng (National Cheng Kung Univ., Tainan, Taiwan)

The present study investigated reliable global measures that can be used to screen/assess infants at-risk for speech problems due to cerebral palsy (CP). Three measures were used in this study: mean duration of utterances, mean duration of syllables, and vocalization rate, which are suggested as likely points of divergence from typical development in infants with CP. Recordings of 9 months and 21 months of ten infants who were at risk of CP and seventeen 9-month-old typically-developing (TD) infants were coded. Naïve listeners filled out a questionnaire about a perceptual evaluation of speech samples of these infants after listening to the first 50 utterances with little background noise in each recording. The questions included, for example, resonance, hoarseness, breathiness, loudness, and tremor. Results of the three measures did not show statistically significant differences between the groups. However, descriptive analyses showed that infants with CP at 9 months produced longer utterances and syllables than when at 21 months. Infants with CP also produced shorter utterances and syllables and fewer vocalizations per minute than TD infants. Furthermore, the correlation between the results of the three measures and perceptual ratings was examined in search for other global measures and to refine the questionnaire questions.

5aSC7. Quantifying the effects of background babble on preschool children's novel word learning in a multi-session paradigm. Meital Avivi-Reich, Megan Y. Roberts, and Tina M. Grieco-Calub (Commun. Sci. and Disorders, Northwestern Univ., Rm. 1-331, Frances Searle Bldg. 2240 Campus Dr., Evanston, IL 60208, m.avivireich@northwestern.edu)

Common daily environments often contain background noise, such as background talkers, that may impose challenges on auditory based learning, including the learning of new words. Extant data on the effects of background noise on novel word learning, however, have shown mixed results. One possible reason for this observation is that most studies assess the effect of background noise in a single session. The present study aimed to add to this body of work by employing a multi-session paradigm to explore the effects of background babble, presented at 0 dB signal-to-noise ratio, on novel word learning in 3-year-old children (n = 8). Children were exposed to two stories presented digitally, each story containing four novel CVC words. Children were exposed to both stories, one in quiet and one in the presence of four-talker babble. After each story, receptive word learning was quantified with a four-alternative-forced-choice task, and expressive word learning was quantified by the number of novel labels correctly produced when their corresponding objects were shown to the children. Results suggested that children's receptive and expressive word learning improved by session; however, greater improvement was observed for the words exposed in quiet. The results and their implications will be further discussed.

5aSC8. Automated episode selection of child continuous speech via blind source extraction. Marisha Speights (Commun. Disord., Auburn Univ., 1199 Haley Ctr., Auburn, AL 36849, mls0096@auburn.edu), Keith D. Gilbert (Speech Technol. & Appl. Res. Corp., Lexington, MA), Joel MacAuslan, and Richard Goldhor (Speech Technol. & Appl. Res. Corp., Bedford, MA)

Commonly, in both research and clinical conditions in which speech recordings are collected from children, the non-targeted adult speech and the targeted child speech are simultaneously recorded during conversational turn-taking tasks, and the utterances of all talkers are audible on every channel of the recording. As a result, the intended analysis of the targeted speaker becomes encumbered by the requirement for a human analyst to identify and mark the episodes in the recording where the targeted speaker is the only active sound source. Blind source extraction (BSE) techniques are utilized here to automatically deliver the isolated child's speech, even when the child's and adult's speech overlap, or when extraneous interfering noise is present. To test the performance of the BSE methods for automated episode selection in child continuous speech, ten typically developing children were each recorded speaking thirty-three sentences in the turn-taking scenario with four microphones placed in the environment. Correlation between the numbers of human-marked episodes identifying the target speakers to the machine-selected episodes was examined. Results of the Pearson correlation indicate there is a significant positive association between hand counts and BSE machine counts ($r = 0.85$, $p < 0.01$).

5aSC9. Perception of talker height and sex from children's voices. Peter F. Assmann, Michelle R. Kapolowicz (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), and Santiago Barreda (Linguist, Univ. of California, Davis, Davis, CA)

Several recent studies have investigated the ability of listeners to judge the height of adult talkers from their speech. Here, we examine the perception of talker height from a sample of /hVd/ syllables spoken by children ranging in age from 5 through 18 years, with 3 talkers of each sex at each age level. Twenty college students judged the height and sex of the talker for each trial. Perceptual estimates of height were moderately correlated with actual height (for males, $r = 0.78$; for females, $r = 0.67$). Consistent with our recent findings concerning the perception of talker age [Barreda & Assmann, JASA-EL 143(5), EL361, 2018], listeners' judgments of talker height appear to incorporate assumptions about talker sex and the relationships between speech acoustics and veridical height for male and female talkers of different heights. For example, when older girls were misidentified as boys, their height was likely to be underestimated. This behavior results in predictable error patterns in height judgments from speech: in situations where talker sex is misidentified, listeners apply the "wrong" acoustic model to estimate talker height, leading to correlated error patterns between apparent height and sex.

5aSC10. Perceptual restoration of interrupted speech in typically-developing school-age children: Effect of interruption rate. Beula M. Magimairaj (Commun. Sci. and Disord., Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72035, bmagimairaj@uca.edu) and Naveen K. Nagaraj (Audiol. and Speech Pathol., Univ. of Arkansas for Medical Sci., Little Rock, AR)

Auditory closure, the ability to fill-in missing information involves bottom-up and top-down influences. There is a lack of systematic studies on children's interrupted speech perception (ISP). We measured 8- to 11-year-old typically developing children's restoration of sentences using the ISP paradigm. We used two interruption rates with noise-filled and unfilled interruptions, respectively. Sentences from the Bamford Kowal Bench

Speech-in-Noise Test were gated with 50% duty cycle square wave at 2.5 and 5 Hz. For the noise-filled condition, silent intervals were filled with speech noise. Preliminary analysis on 33 children has been completed and 40 more children are scheduled complete participation. Accuracy of keyword recognition was subjected to 2 (interruption rate: 2.5 vs. 5 Hz) X 2 (noise-filled vs unfilled) repeated measures ANOVA. Overall, main effect of interruption rate, $F(1, 32) = 500$, $p < 0.0001$ and filler noise was significant, $F(1, 32) = 289$, $p < .00001$. Recognition scores for noise-filled conditions were better than unfilled for both 2.5 Hz (mean difference = 27.2) and 5 Hz (mean difference = 5.03) conditions. Larger perceptual restoration at 2.5 Hz compared to 5 Hz may indicate greater reliance on lexical knowledge at 2.5 Hz to restore speech than on phonemic restoration.

5aSC11. Children's susceptibility to irrelevant speech on a selective attention task: What role does working memory play? Naveen K. Nagaraj (Audiol. and Speech Pathol., Univ. of Arkansas for Medical Sci., 2801 S. University, Suit 600, Little Rock, AR 72204, nnagaraj@uams.edu) and Beula M. Magimairaj (Commun. Sci. and Disord., Univ. of Central Arkansas, Conway, AR)

Irrelevant speech effect (ISE) is well studied in adults; however, limited studies have explored ISE in children. We examined individual differences in working memory (WM) capacity on ISE due to proactive interference and interference from the irrelevant channel using a dichotic selective attention task. Eighty-six 7- to 11-year-old children were tested. Two streams of digits were presented (to-be-ignored and to-be-attended) to investigate both proactive and irrelevant channel interference in quiet and in speech babble. The irrelevant channel interference was estimated by tracking the response from the to-be-ignored ear. Proactive interference corresponded to participants reporting previously presented stimuli. Children's WM capacity was measured using the Woodcock Johnson Auditory WM task and an experimental complex memory span task. Based on the attention control theory of WM, it was predicted that both irrelevant channel interference and proactive interference would be negatively related to children WM capacity. Results showed no such correlation between WM capacity and ISE, which may have been due to the inverse relation between the two types of interference. This is because proportion of total errors that corresponded to proactive interference was positively correlated with WM, whereas irrelevant channel interference was negatively correlated with children's WM capacity.

5aSC12. Advancing clinical group verification framework for screening child speech sound disorders using "text-independent" i-Vectors. Prasanna V. Kothalkar (Jonsson School of Eng. and Comput. Sci., Ctr. for Robust Speech Systems, UT Dallas, 800 W Campbell Rd., Richardson, TX 75080-2277, prasanna.kothalkar@utdallas.edu), Johanna M. Rudolph, Christine Dollaghan, Jennifer McGlothlin, Thomas F. Campbell (Commun. Disord., Univ. of Texas at Dallas, Dallas, TX), and John H. L. Hansen (Jonsson School of Eng. and Comput. Sci., Ctr. for Robust Speech Systems, UT Dallas, Richardson, TX)

I-Vectors are the current state-of-the-art feature representation in acoustic event identification tasks such as speaker recognition, language recognition, etc. They are referred to as identity vectors since they represent a unique quality of the speaker and have been useful for detecting adult speech pathologies . Speech Sound Disorders (SSDs) affect between 3% and 16% of US children and are difficult to detect due to the presence of developmental speech sound errors. We have been working to automate the process of speech screening. Our dataset consists of 29, single word recordings from 165, 3–6 year old children and was collected using an iOS application. Children were assigned to clinical groups using a percentage consonants correct growth curve model. Sixty-four children were classified as exhibiting an SSD, the rest as exhibiting normal speech acquisition. To achieve our purpose, we first introduced our clinical group verification framework using Gaussian Mixture Models. We extended the framework to screen the children's speech based on single words using "text-dependent" i-Vectors, along with L2-logistic regression and Gaussian backend machine learning classifiers. This improved the algorithms' accuracy. In the current study, we modified our offline post-processing algorithms within the framework, which provided excellent results for "text-independent" i-Vectors. Apart from the algorithms, we also present detailed visual analysis of the classifier score transformation, during the post-processing phase.

5aSC13. Velopharyngeal control for speech in children with cochlear implants: Nasalance data in vowel and consonant segments. Laura L. Koenig (Haskins Labs and Adelphi Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu), Areti Okalidou (Univ. of Macedonia, Salonika, Greece), and George Psillas (Dept. of Medicine, Aristotle Univ. of Thessaloniki, 1st Univ. ENT Clinic of Ahepa Hospital, Thessaloniki, Greece)

Children with hearing loss before the development of their phonological system display various kinds of atypical speech production characteristics. Cochlear implants (CIs) can lead to improved speech production, although children may vary considerably in their outcomes. Past work suggests that one effect of hearing impairment is poor control over the velopharyngeal system, but little direct data exist to show how CIs improve velopharyngeal function for speech. This work presents nasometer data for Greek-speaking children with CIs, ages 4–16 years, along with age- and gender-matched typically-hearing children. Children produced bisyllabic words varying in stress position and in the type of initial stop consonant: Nasal, voiceless unaspirated, and voiced (which, in Greek, may be produced as prenasalized). The nasal and oral microphone signals from the nasometer were labelled in Praat to demarcate consonantal and vocalic regions. Preliminary analyses on the whole words suggested that children with CIs are comparable to hearing controls for words beginning with nasal consonants, but differed for words with oral consonants. This presentation will present durations and average nasalance over segmental (consonantal and vocalic) regions, permitting more fine-grained assessment of velopharyngeal control.

5aSC14. Quantifying expressive word learning in the presence of background speech. Katherine M. Simeon, Katie Schramm (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Frances Searle Bldg. Rm. 2-381, Evanston, IL 60208, ksimeon@u.northwestern.edu), Katherine Gordon (Childhood Deafness, Lang. and Learning, Boystown National Res. Hospital, Omaha, NE), and Tina M. Grieco-Calub (Commun. Sci. & Disord., Northwestern Univ., Evanston, IL)

Children acquire novel word-object pairs rapidly and with few exposures via *fast-mapping* (Carey & Bartlett, 1978). However, everyday environments contain background noise that interferes with children's formation of phonological representations of words. Existing research has not consistently shown disruption of receptive word learning by background noise when implementing a closed-set, forced-choice paradigm. In contrast, Riley & McGregor (2012) found that background noise disrupted expressive word form representation. The present study extends this prior work by investigating the effects of background speech on expressive word learning in preschool-aged children. Three-to-four-year-old children performed a fast-mapping task in quiet and in the presence of two-talker speech presented at a +2 dB signal-to-noise ratio. Children viewed short animations whereby novel objects were verbally labeled. Children were then tested on each novel label-object pair with a tiered expressive recall task (Gordon & McGregor, 2014). For each object, children were asked to verbally produce its novel label. If children were unable to name the object, they were prompted with the label-initial sound (e.g., cueing) or given a closed-set of potential labels from which to choose the correct one. This presentation will discuss how depth of expressive word learning can vary in quiet and noisy environments.

5aSC15. Children's ratings of vocal emotion intensity depend on the emotion spoken and speaker familiarity but not acoustic parameters. Tawni B. Stoop, Peter Moriarty, Michelle Vigeant, Rick Gilmore (Penn State Univ., State College, PA), Pan Liu (Penn State Univ., Dept. of Psych., Western Interdisciplinary Res. Bldg., Rm. 2172, London, ON N6A 5B7, Canada, pliu261@gmail.com), and Pamela Cole (Penn State Univ., State College, PA)

Scherer (1986) documented the acoustic parameters associated with adults' perceptions of discrete vocal emotions. We investigated physical and psychological factors that influence children's ratings of discrete vocal emotions using stimuli that approximated naturally occurring speech, including familiarity with the speaker. Specifically, we presented 52 7- and 8-year-olds with one side of a brief phone conversation spoken in happy, angry, sad, and non-emotional prosodies by both the child's mother and another child's mother, unfamiliar to the target child. As a group, the familiar and unfamiliar mothers' prosodies did not differ in fundamental frequency (F0), F0 standard deviation, or speech rate—acoustic parameters that are most identified with angry, happy, and sad prosodies. Children accurately recognized the emotion spoken: They rated angry stimuli as more angry than happy or sad. Regression analyses indicated that speaker familiarity predicted children's intensity ratings even after the target acoustic parameters were taken into account, but this effect was moderated by speaker emotion, such that children rated their mothers as more intensely angry than unfamiliar mothers. The findings suggest that their mother's angry voice holds psychological significance for children that is not explained by variations in its most salient acoustical properties.

5aSC16. Perception-production relations in residual speech errors before and after biofeedback training. Heather M. Campbell, Laine Cialdella, and Tara McAllister Byun (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., Fl. 6, New York, NY 10012, heather.campbell@nyu.edu)

Rhotics are among the latest-emerging, most frequently misarticulated sounds in acquisition of American English (Smit *et al*., 1990). Children with residual speech errors (RSE) affecting rhotics may be unable to perceive differences between their own accurate and inaccurate productions (Shuster, 1998). Visual biofeedback may be an efficacious treatment approach for these children because it can bypass deficient auditory-perceptual channels (Shuster *et al*., 1995). Previous research has shown a relationship between accuracy in perceiving and producing rhotic targets in typically developing (TD) children (McAllister Byun & Tiede, 2017). We used the same methods to measure rhotic perception and production in a sample of 75 children with RSE. Based on previous literature (e.g., Hearnshaw *et al*., 2018; Rvachew & Jamieson, 1989), we hypothesize that children with RSE will show poorer perceptual abilities on average than TD children, but that they will show a similar degree of association between perception and production. Second, we will test whether changes in auditory acuity take place after a period of biofeedback treatment for RSE. Finding such changes would suggest that biofeedback may have its effect not only by retraining a target motor plan, but also by helping children update their auditory targets.

5aSC17. Perseverance measured in children's oral reading. Jared Bernstein, Jian Cheng (Analytic Measures Inc., 1330 Tasso St., Palo Alto, CA 94301, jared413@stanford.edu), and Ahmed Magooda (Comput. Sci., Univ. of Pittsburgh, Pittsburgh, PA)

Do struggling readers score lower on reading tests in part because they get discouraged or slow down? Schools track reading progress in grades K-4 by measuring oral reading fluency (ORF) as students read aloud. The usual ORF measure is accurate reading rate, which is reported in words correct per minute (WCPM). Computer recognition and processing of speech has automated the scoring of oral reading and enabled new analyses of reading performance, particularly concurrent scoring of shorter text units such as paragraphs, sentences, and words that occur within a passage that is read for comprehension. Do stronger readers perform more consistently within a passage and across passages during a test, while stuggling readers do not persevere. If true, then some conventional reading tests may conflate perseverance with basic reading skills. We have read-aloud data from 600 students in grades 1–5, but now only compare quintile 2 to quintile 4 in a sample of 100 fourth grade students. We find among both reader groups that passage performance tends not to diminish across passages late in a test, but remains roughly constant. However, within each text passage, the fast readers tend to slow down and the slower readers tend to speed up.

5aSC18. Speech accommodation across communication partners by adolescent speakers. Kathryn Connaghan and Zachary Kopp (Commun. Sci. & Disord., Northeastern Univ., 360 Huntington Ave., Forsyth Bldg. - Rm. 226, Boston, MA 02115, kconnaghan@mghihp.edu)

Communication context has considerable impact on speech production, with speakers accommodating their speech based on the needs and attributes of their communication partner. For instance, speakers demonstrate acoustic-phonetic convergence, wherein they alter their speech characteristics to converge with those of their communication partner (e.g., Borrie & Liss, 2014). This speech modulation promotes closeness, shared identity, and communication effectiveness. While documented in adult speakers, the occurrence and presentation of speech accommodation throughout development is unclear. The current study compared the acoustic characteristics of speech samples produced by adolescent speakers in isolation versus in response to communication partners. Twelve neurotypical male adolescents (13–17 years) read sentences aloud with and without pre-exposure to speech recordings collected from four speakers varying by age and sex. Acoustic correlates of rhythm and prosody were compared across elicitation conditions. As a group, the participants significantly modulated their peak fundamental frequency and intensity across some conditions. Convergence was evaluated with bivariate correlations between exposure and response utterances. The highest correlation was observed for speech rate with the adult male, though unique patterns of convergence were observed across participants. Further study of speech accommodation by individuals with speech disorders may inform interventions to support successful social interactions.

5aSC19. Vocal changes across disease progression in amyotrophic lateral sclerosis (ALS). Jordan Green, Kathryn Connaghan (Commun. Sci. & Disord., MGH Inst. of Health Professions, 36 First Ave., Boston, MA), (kconnaghan@mghihp.edu), Yana Yunusova (Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada), Kaila Stipancic (Commun. Sci. & Disord., MGH Inst. of Health Professions, Boston, MA), Sarah Gutz (Harvard Univ., Boston, MA), and James Berry (Massachusetts General Hospital, Boston, MA)

Amyotrophic lateral sclerosis (ALS) is a neurodegenerative disease characterized by loss of muscle strength and function. The speech systems (respiratory, phonatory, velopharyngeal, and articulatory) are frequently affected, causing speech and swallowing impairments. Changes to voice production are often reported. These changes include altered fundamental frequency, phonatory instability, and the development of breathy or harsh voice quality (see Green *et al*., 2013 for review). While acoustic studies of speakers with ALS have demonstrated voice dysfunction, the findings have been variable in the type and direction of change across individual speakers. The current investigation seeks to further explicate vocal dysfunction and change in ALS by exploring promising acoustic phonatory measures across disease progression. Participants with and without ALS were audiorecorded at multiple time points while producing sustained vowels and connected speech. Acoustic analyses include traditional phonatory measures extracted from vowel prolongations (fundamental frequency, jitter, and shimmer) and newer measures of phonation across connected speech (e.g., cepstral peak prominence). Voice metrics are compared between speaker groups and across time points. It is anticipated that the findings will support the development of valid and reliable measures to mark disease onset and progression, thereby facilitating intervention and mitigating the devastating impact of ALS.

5aSC20. Fricative productions of Mandarin-speaking children with cerebral palsy: The case of five-year-olds. Chin-Ting Liu (Lang. Ctr., National Chin-Yi Univ. of Technol., Taichung City, Taiwan), Li-mei Chen (National Cheng Kung Univ., Tainan, Taiwan), Katherine C. Hustad, Ray D. Kent (Univ. of Wisconsin - Madison, Madison, WI), and Chia-Cheng Lee (Portland State Univ., 1831 SW Park St., Apt. 112, Portland, OR 97201, cl9@pdx.edu)

The age means between cerebral palsy children (CPs) and typically developing children (TDs) were not ideally matched in previous studies targeting the fricative productions of Mandarin-acquiring CPs (Liu *et al*., 2017, 2016). Therefore, in this study, age-matched CPs (N = 8, Mean = 5;6) and TDs (N = 8, Mean = 5;6) were recruited. Six acoustical parameters (frication duration, rise time, and first to fourth spectral moments) of the five Mandarin Chinese voiceless fricatives (labiodental, alveolar, alveolo-palatal, velar, and retroflex) produced by the participants were compared. The results from Mann-Whitney U tests indicated that, for the labiodental, alveolar, alveolo-palatal, and retroflex fricatives, the values from frication duration, rise time, and first and second spectral moments produced by CPs were generally significantly lower than those produced by TDs. The current results from age-matched groups were different from previous studies and suggested that: 1) CPs tended to have a more posterior constriction site and a more restricted lingual constriction area with the passive articulator; 2) the third and fourth spectral moments were less effective in quantifying the differences between the two groups. Future studies are suggested to observe the fricative productions from older or younger age-matched groups so that the fricative developmental trajectory of Mandarin-acquiring CPs can be faithfully depicted.

5aSC21. Acoustic analysis of the speech of patients with oral cancer: A methodological investigation comparing speech stimuli derived from different clinical outcome measures. Agnieszka Dzioba (Surgery, Univ. of AB, 8440 112 St., Edmonton, ON T6G 2BT, Canada, adzioba0@gmail.com), Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada), Hadi Seikaly (Surgery, Univ. of AB, Edmonton, AB, Canada), and Jana Rieger (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

Background: Oral cavity cancer and its treatment reduce the quality of speech. Clinicians often utilize a variety of outcomes measures to assess speech function of patients with oral cavity cancer reducing the comparability of formant results due to differences in linguistics and phonetic contexts. **Objective:** To evaluate the degree of agreement in vowel space size in a F1-F2 plane, when comparing speech stimuli selected from a controlled environment (i.e., /hVd/ context) to speech stimuli segmented from two clinically available speech outcome measures in patients treated for cancer of the oral tongue. *Methods:* Voice recordings of nine patients treated with primary surgery for cancer of the oral cavity who attended functional assessment appointments (pre-operatively, and at 1-, 6-, and 12-months post-operatively) were analyzed. Agreement between vowel space size obtained from /hVd/ phrases and other speech assessments were compared using linear correlations and t-tests. *Results:* Vowel space size derived from /hVd/ phrases correlates strongly with corresponding estimates from other speech tasks ($r = 0.72$ to 0.82, $p < 0.0001$). The other tasks had significantly reduced vowel space sizes compared to /hVd/ ($t = 4.7$ to 5, $p < 0.0001$). *Conclusion:* Vowel space size estimates based on different speech tasks are mutually comparable and may offer insight to oral cancer speech production acoustics.

5aSC22. Tongue position during speech as a function of palatal anatomy: Controls and glossectomy patients. Maureen Stone, Elise Tigani, Elise Tigani (Univ. of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Baltimore, MD 21201, mstone@umaryland.edu), Junghoon Lee (Johns Hopkins Medical Institutions, Baltimore, MD), Jonghye Woo (MGH, Boston, MA), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD)

This study examined the effect of palatal anatomy on tongue position during the complex speech movement seen in the motion from /ʃ/ to /l/ in "shell." The study was interested in whether subjects with flatter/narrower palates would use more tongue anteriority than those with higher/wider palates. Anteriority is the percent of the tongue anterior to the first molar. Low palate speakers may use more anteriority due to less space in the palatal vault. The speech task "a shell" requires complicated deformation between the /ʃ/, which elevates the sides of the tongue, and the /l/, which lowers the sides, elevates the tongue tip, and retracts the root. In the past, palate effects have been shown to explain idiosyncratic subject differences. Palate height explained the use of apical vs laminal /s/. Palate width affected the anteriority of tongue position in cancer patients after glossectomy surgery. Therefore, this study examined 16 controls and 8 glossectomy patients. Initial measurements show more anteriority for patients than controls. In addition, patients with a large overbite (vertical overlap between upper and lower incisor) had more anteriority due to a smaller front cavity.

5aSC23. The acoustic consequences for sibilants produced by glossectomies compared to healthy controls. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), Maureen Stone (Neural & Pain Sci. and Orthodontics & Pediatrics, Univ. Maryland School of Dentistry, Baltimore, MD), and Wei-Rong Chen (Haskins Labs., New Haven, CT)

Glossectomy surgery affects the ability to elevate the tongue tip (Grimm *et al*, JSLHR v.60, 3417–3425, 2017), which is known to affect production of /s/. Interestingly, /ʃ/ is less affected by glossectomy surgery. In this study, the acoustics of normal and aberrant /s/ and /ʃ/ were characterized using three parameters developed for normal adults and adolescents. Main spectral peak (F_M) indicates place of constriction. Amplitude difference (A_D) indicates degree of sibilance. LevelD indicates relative energy between high and mid-frequencies, reflecting source changes. Subjects were two patients and three healthy controls for whom acoustic, perceptual, and MRI data existed. The controls were matched by gender and vocal tract size to the patients. The corpus included three words and four VCV syllables where C was /s, ʃ/. Results showed the male patient, whose /s/ tokens were perceived as /ʃ/, had low F_M values for /s/. The female patient had more varied and less severe errors. She produced perceptually-distinct /s/ and /ʃ/ and showed normal means but larger variances for /s/ than /ʃʃ/ parameters. Overall, the acoustic parameters match both the perceptual and articulatory (MRI) results, and help to explain the detailed effects of glossectomy surgery on the post-surgery productions. [Work supported by NIH CA-133015.]

5aSC24. Diphthong productions in speakers with Parkinson's disease across speaking modes. Yunjung Kim (Louisiana State Univ., 86 Hatcher Hall, Baton Rouge, LA 70803, ykim6@lsu.edu) and Austin R. Thompson (Louisiana State Univ., Baton Rouge, LA)

Several vowel acoustics parameters (e.g., second formant frequency (F2) slope, vowel space area) are known to significantly contribute to speech intelligibility in dysarthria (Kim, Kent, Weismer, 2011; Lansford & Liss, 2014). Particularly, F2 slope has been identified as an important predictor of speech intelligibility regardless of underlying neuropathologies of dysarthria (Kim, Weismer, Kent & Duffy, 2009). The current study focuses on intra-speaker modifications of F2 slope in speakers with Parkinson's disease (PD) voluntarily elicited across three speaking modes, less clear, conversational, and more clear speech, in recognition that prior work has mostly focused on inter-speaker comparisons. Varying speech clarity was our interest, given its wide use as an efficient behavioral therapeutic approach in clinic. Twenty speakers (10 PD, 10 neurologically healthy) were asked to read diphthongs embedded in real words. Diphthongs were chosen because they require a significant change in configuration of vocal tract and, therefore, are sensitive to the presence/severity of dysarthria. In the presentation, distributions of F2 slope 1) between PD and healthy speakers, and 2) across speaking modes within each speaker will be discussed. In addition, articulatory data will be presented for part of the speakers from whom kinematic data were obtained using the electromagnetic articulography (EMA) system.

5aSC25. Enhancing speech adaptation to a palatal perturbation using ultrasound visual biofeedback. Guillaume Barbier (Université de Montréal, Université de Montréal, École d'orthophonie et d'audiologie, C.P. 6128, succursale Centre-ville, Montreal, QC H3C 3J7, Canada, barbier_guillaume@orange.fr) and Douglas Shiller (Université de Montréal, Montreal, QC, Canada)

Prior clinical studies suggest that visual biofeedback of the tongue (e.g., ultrasound) may enhance the treatment of speech disorders, but outcomes have been mixed, likely due to variability in both the clinical profiles of participants and the way in which treatments have been carried out. Understanding the true potential of this clinical tool requires a clearer

sense of how visual biofeedback interacts with speech motor control. Here, we present a novel experimental approach that mimics the conditions of clinical treatment for speech disorders while maintaining a high level of consistency and control over the speech-learning task. The procedure involves altering the vocal tract of typically developing talkers using a palatal prosthesis to perturb /s/-production in combination with the controlled application of ultrasound biofeedback. As participants practice and improve their speech, changes in articulatory movements are examined using acoustic and kinematic measures. The present study compared a control group (n = 10) receiving only auditory feedback during speech practice with a group receiving visual biofeedback (n = 10). Results indicate an effect of biofeedback, in particular in the retention of learned motor patterns, indicating that talkers integrate real-time visual feedback of tongue movement into the sensorimotor processes driving speech adaptation.

5aSC26. The impact of melodic intonation therapy on rate and fundamental frequency in a client with Palilalia. Christina C. Akbari (Commun. Disord., Arkansas State Univ., P.O. Box 910, State University, AR 72467, cakbari@astate.edu)

Palilalia is a rarely documented speech disorder associated with Parkinson's disease, stroke, epilepsy, etc. Palilalia is characterized by an increased rate of speech along with repetitions of words, phrases, or sentences. Only one published article has discussed treatment of Palilalia. Helm (1979) utilized a pacing board to help his 54-year-old client speak slower and reduce his repetitions. The participant touched the divided board with his fingers as he produced each syllable. This board reportedly slowed the patient's speech. Melodic intonation therapy (MIT) was designed to help individuals with aphasia communicate more effectively (Sparks & Holland, 1976). MIT involves intoning propositional phrases and sentences using a limited range of musical notes to produce utterances along with tapping to keep the rhythm. Speech productions with MIT resulted in slower, more lyrical productions in comparison to spontaneous speech (Sparks & Holland). MIT incorporates aspects of pacing and rhythm that could be effective in slowing speech rate. This study utilized a single-subject design involving a 42-year-old male who experienced an anoxic brain injury. Speech samples from therapeutic sessions were collected and analyzed. The results indicated significant differences in terms of rate and fundamental frequency between spontaneous utterances and those produced using MIT strategies.

5aSC27. Predicting scores on the Montreal cognitive assessment from a spontaneous speech sample. Alan A. Wisler, Annalise Fletcher, and Megan J. McAuliffe (New Zealand Inst. of Lang., Brain and Behaviour, NZILBB, Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, alanwisler@gmail.com)

This presentation examines acoustic and linguistic measurements taken from 521 older participants. The aim was to examine the relationship between measurements derived from spontaneous speech and participants' scores on the Montreal Cognitive Assessment (MOCA). Though both speech measurements and MOCA scores have been shown to be predictive of mild cognitive impairment, there has been little exploration in the literature of the relationship between these two domains. Participants completed the MOCA assessment and were prompted to describe an early childhood memory. The lexical complexity analyzer toolkit was used to extract a range of measurements to assess the lexical density, sophistication, and variation in each participant's storytelling passage. For the acoustic analysis, we drew a subset of the features from the Interspeech 2010 Paralinguistic Challenge set and included measures of duration and speaking rate. A least absolute shrinkage and selection operator (LASSO) approach was used to model the relationship between acoustic, lexical, and demographic information and participants' MOCA scores. Using the covariance test statistic we identified six important variables: one lexical, three acoustic and two demographic. This reduced model explained 18.06% of the variance in participants' MOCA scores. This R^2 value was validated on a held out set of 121 participants.

5aSC28. The effect of noise on the intelligibility of electrolaryngeal speech. Kimberly McNicholl, Steven R. Cox, Alexandra Chill (Commun. Sci. and Disord., Adelphi Univ., Hy Weinberg Ctr. 136, Garden City, NY 11530), (scox@adelphi.edu)

The electrolarynx (EL) is a hand-held device used for voice and speech production following surgical removal of the larynx. Many individuals who have their larynx removed begin using an EL in the immediate, postsurgical period. Unfortunately, EL speech is known to be poorly understood in quiet listening conditions when compared to laryngeal and alaryngeal speech methods. However, there is a dearth of research studying the effects of background noise and its impact on the intelligibility of EL speech. The purpose of this study, then, was to investigate how well naïve listeners identify EL speech in quiet and noise conditions. Ten EL speakers were recorded while reading 10 sentences containing five keywords per sentence. Twenty listeners were asked to orthographically transcribe 50 sentences each while listening to the EL speakers' recordings. Preliminary data indicate that listeners identified 76% of keywords (*Mdn* = 78%; *SD* = 5%, *range*= 69–80%) in quiet compared to 60% of keywords (*Mdn* = 63%; *SD* = 10%; *range* = 44–69%) in noise. Implications and future directions will be discussed.

5aSC29. Visual-aerotactile perception and congenital hearing loss. Charlene Chang (Speech and Audiol. Sci., Univ. of Br. Columbia, 2177 Wesbrook Mall, Vancouver, BC, Canada, charlenechang94@gmail.com), Megan Keough, Murray Schellenberg, and Bryan Gick (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Previous research on multimodal speech perception with hearing-impaired individuals focused on audiovisual integration with mixed results. Cochlear-implant users integrate audiovisual cues better than perceivers with normal hearing when perceiving congruent [Rouger *et al.* 2007, PNAS, 104(17), 7295–7300] but not incongruent cross-modal cues [Rouger *et al.* 2008, Brain Research 1188, 87–99), leading to the suggestion that early auditory exposure is required for typical speech integration processes to develop (Schorr 2005, PNAS, 102(51), 18748–18750). If a deficit of one modality does indeed lead to a deficit in multimodal processing, then hard of hearing perceivers should show different patterns of integration in other modality pairings. The current study builds on research showing that gentle puffs of air on the skin can push individuals with normal hearing to perceive silent bilabial articulations as aspirated. We report on a visual-aerotactile perception task comparing individuals with congenital hearing loss to those with normal hearing. Results indicate that aerotactile information facilitated identification of /pa/ for all participants ($p < 0.001$) and we found no significant difference between the two groups (normal hearing and congenital hearing loss). This suggests that typical multi-modal speech perception does not require access to all modalities from birth. [Funded by NIH.]

5aSC30. Testing and evaluation paradigms for assistive hearing devices. Ram Charan M. Chandra Shekar (Cochlear Implant Lab. – Ctr. for Robust Speech Systems (CRSS), Dept. of Elec. Eng., Erik Jonsson School of Eng. and Comput. Sci., Univ. of Texas at Dallas, 7780 McCallum Blvd., Apt. No 26307, Dallas, TX 75252, RamCharan.ChandraShekar@utdallas.edu), Hussnain Ali, and John H. L. Hansen (Cochlear Implant Lab. – Ctr. for Robust Speech Systems (CRSS), Dept. of Elec. Eng., Erik Jonsson School of Eng. and Comput. Sci., Univ. of Texas at Dallas, Richardson, TX)

Research interfaces and assistive hearing devices (RI/AHD) enable investigators to perform perceptual studies and develop effective sound processing strategies for cochlear implant (CI) and hearing aid (HA) devices. Several concerns exist regarding safe and reliable operation of RI/AHDs, for example, safety, reliability, possible hazards, accuracy, and their long-term consistency. Recently, CRSS-CILab at UTDallas has developed a mobile RI (CCi-MOBILE) for unilateral and bilateral HA/CI devices. In this study, a comprehensive testing and evaluation strategy is proposed to investigate the behavior of RI/AHDs that addresses safety concerns, reliability of hardware/firmware design, hearing-comfort, subjective sound-quality, and range of human-driven settings, as well as diverse types of acoustic exposure. Furthermore, a characteristic risk-hazard analysis is performed including diagnosis

and categorization of the severity level of the associated safety concerns by probing pulse characteristics from each electrode, e.g., current-level, pulse-width, charge per phase, frame-period (for stimulation-rate), inter-phase gap (IPG), and several other parameters. The CCi-MOBILE RI is employed in this study to demonstrate the utility of the proposed testing and evaluation paradigm. The described practices could potentially serve as a blue-print to characterize future RI for AHDs and several other personalized listening devices based on safety, reliability, listening comfort/perception, and subjective sound quality.

5aSC31. Salience of cochlear implant users' speech rate. Valerie Freeman (Dept. of Commun. Sci. and Disord., Oklahoma State Univ., 042 Murray Hall, Stillwater, OK 74078, valerie.freeman@okstate.edu)

Speech rate-matching is a form of rapid accommodation in which speakers adapt their speech rates to match their interlocutor's previous utterance. Such responsiveness may contribute to rhythmic convergence between speakers throughout a conversation. However, little is known about interactions between typical speakers and those with speech or hearing difficulties. In recent work, prelingually deaf cochlear implant (CI) users were poorer rate-matchers than their peers, but it was unclear whether interlocutors accommodated toward CI users' speech rates, which can vary widely between individuals. A follow-up study adapted procedures from work in which participants rate-matched toward fast- and slow-talking Parkinson's patients. Surprisingly, people did not rate-match to either CI users or controls. This study explores one possible explanation: that differences in speech rate were not salient enough for participants to modify their own rates in response. Following previous procedures, participants (a) alternated hearing CI users' sentences and reading other sentences, (b) rated CI users' utterances as fast, slow, or neither, and (c) repeated the first task with different stimuli. Results will show whether participants were able to identify differences in speech rate (when prompted) and whether they improved in speech rate-matching after speech rate was brought to their attention.

FRIDAY MORNING, 9 NOVEMBER 2018 RATTENBURY A/B (FE), 9:00 A.M. TO 11:45 A.M.

Session 5aSP

Signal Processing in Acoustics: General Topics in Signal Processing II

Edward Richards, Cochair
Scripps Oceanography, University of California, San Diego, 8820 Shellback Way, Mail Code 0238, La Jolla, CA 92093

Michael J. Bianco, Cochair
Marine Physical Laboratory, University of California San Diego, Scripps Institution of Oceanography, 9500 Gilman Dr., La Jolla, CA 92037

Contributed Papers

9:00

5aSP1. Mathematical properties of generalized Bessel functions and application to multi-tone sinusoidal frequency modulation. Parker Kuklinski (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, parker.s.kuklinski1@navy.mil) and David A. Hague (Naval Undersea Warfare Ctr., North Dartmouth, MA)

The generalized Bessel function (GBF) is a multi-dimensional extension of the standard Bessel function. Two dimensional GBFs have been studied extensively in the literature and have found application in laser physics, crystallography, and electromagnetics. However, a more rigorous treatment of higher-dimensional GBFs is lacking. The GBF exhibits a rich array mathematical structure in regards to its partial differential equation representation, its asymptotic characterization, and its level sets. In this talk/paper, we explore these properties and connect spectral and ambiguity function optimization of a multi-tone SFM signal to finding the location of the roots of these generalized Bessel functions.

9:15

5aSP2. Two-dimensional high-resolution acoustic localization of distributed coherent sources for structural health monitoring. Tyler J. Flynn and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

Many high-resolution techniques exist for localizing acoustic sources; however, most of these methods suffer when applied to data containing spatially separated coherent signals. The Spectral Estimation Method with Additive Noise (SEMWAN) is an existing beamforming technique for high-resolution localization of incoherent monopole sources in low SNR environments. SEMWAN utilizes a reference measurement to subtract, and thereby suppress, background noise, but its performance suffers for cases involving mutually-coherent sources—such as a distributed radiating source. SEMWAN has previously been extended to one-dimensional coherent source scenarios by implementing the additional step of subarray averaging. This extra processing step

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reduces *coherent* background signatures and thereby permits localization of small acoustic changes (e.g., localized damage) provided a known baseline array recording is available. This presentation discusses the extension of this approach to two-dimensional problems using Cartesian planar arrays. Experimental results using multiple coherent sources and a remote receiver array (with varying numbers of elements and subarray geometry) at frequencies from 1.0 to 10 kHz are analyzed and compared to simulation results. Application of this technique for localization of mechanical changes in a 30-cm-square aluminum plate is also addressed. [Sponsored by NAVSEA through the NEEC and by the US DoD through an NDSEG Fellowship.]

9:30

5aSP3. Explorations of *in-situ* source localization in the deep ocean using frequency difference matched field processing and matched autoproduct 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 (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched field processing (MFP) is a well-known technique for source localization in challenging environments. MFP involves correlating array-recorded fields with calculated replica fields developed from knowledge of the acoustic environment. Incomplete knowledge of the environment causes mismatch between measured and calculated fields, and this mismatch can cause MFP to fail at relevant ranges and frequencies in the deep ocean. Given that the severity of this mismatch increases with frequency, a proposed remedy to the mismatch problem is to analyze the frequency-difference autoproduct of the recorded field, rather than the field itself. The phase of the frequency-difference autoproduct is expected to mimic that of an out-of-band field at a selectable below-band frequency, thereby mitigating the severity of the mismatch at in-band frequencies. This autoproduct is then matched either to a replica field at the below-band frequency, or to an appropriate calculated autoproduct field. This presentation explores these methods, and higher-order corrections, to localize moored sources using data from the North Pacific Acoustic Library. Localization statistics are presented based on the closest moored source, 130 km from the receiving array with a signal bandwidth from 200 to 300 Hz. [Sponsored by ONR.]

9:45

5aSP4. Using coherence to improve calculation of active acoustic intensity. Mylan R. Cook, Joseph S. Lawrence, Kent L. Gee, Scott D. Sommerfeldt, Tracianne B. Neilsen, and Michael C. Mortenson (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

The Phase and Amplitude Gradient Estimator (PAGE) method was developed as an alternative to the traditional method for calculating energy-based acoustic measures such as active acoustic intensity [E. B. Whiting *et al*. J. Acoust. Soc. Am. **142**, 2208–2218 (2017)]. While this method shows many marked improvements over the traditional method, such as a higher valid frequency bandwidth for broadband sources, contaminating noise can lead to inaccurate results. Both the traditional method and the PAGE method can perform poorly when microphone pairs exhibit low coherence, whether caused by noise or by imprecise measurements. By using a coherence weighting in the least-squares pressure gradient calculations, where the weighting can vary across frequency, better estimates of the pressure gradient can be obtained, which in turn yield more accurate results for intensity. Additionally, a coherence-based approach may be used to mitigate the negative impact of contaminating noise, most especially for uncorrelated contaminating noise. These improvements, though requiring a greater amount of computation time, can be integrated into the PAGE method processing procedures to create a more stable method for calculating acoustic intensity. [Work supported by NSF.]

10:00

5aSP5. Application of time-domain near-field acoustical holography to the high-order ambisonics presentation of moving sound sources. 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)

High-order ambisonics (HOA) is a sound field reproduction method based on results from near-field acoustical holography (NAH). As such, it is formulated in the frequency domain; this assumes a stationary acoustic field. This presentation applies a recent time-domain formulation of NAH by Attendu and Ross. The approach is based on the linear deconvolution along the spatial coordinates and samples the Green's function in the time and spatial domains, as opposed to the Fourier domain. Our contribution is an application of these ideas to overcome the problems inherent in the HOA reproduction of non-stationary sound fields, in particular those of moving sound sources. In practice, the rendering of moving sound sources is approximated by assuming they are stationary within short temporal windows. The approximation, while useful for slow-moving sources, cannot recreate the Doppler shift and results in discontinuities along the temporal structure of the sound field. On the other hand, the proposed method does not impose any constraints on the motion of the sound sources. Unlike conventional HOA, the proposal cannot be formulated as a simple filter bank; however, numerical experiments show that fast moving sources exhibit the expected Doppler shift.

10:15–10:30 Break

10:30

5aSP6. Resolving range aliasing in Omega-K synthetic aperture beamforming. Timothy Marston (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

The Omega-K algorithm is a beamforming algorithm that exploits Stolt migration in the wavenumber domain to focus stripmap synthetic aperture data. Because of its efficiency and its inherently broad-band properties it has been a common solution for synthetic aperture beamforming problems, especially in situations where trajectory perturbations are small and computational resources are limited. The algorithm is commonly divided into two stages: a bulk compression stage leveraging a phase multiplication in the wavenumber domain (sometimes used in isolation for resonance and shadow enhancement), which focuses data at the scene center, and Stolt migration which brings the entire scene into focus. For systems with very broad beamwidth the bulk compression stage can lead to spatial aliasing for close range scatterers, causing a loss of spatial spectrum information and increased noise. The cause of this frequently overlooked aliasing problem, its manifestation in SAS images, and methods for its avoidance will be discussed and demonstrated using both simulations and ClutterEX17 sonar data.

10:45

5aSP7. Designing waveforms with desired ambiguity function mainlobe and sidelobe structure. David A. Hague (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, david.a.hague@gmail.com)

In active sonar, a transmit waveform's ability to resolve closely spaced targets in range/Doppler and distinguish weak targets in the presence of a stronger one are determined by the mainlobe and sidelobe structure, respectively, of the waveform's broadband ambiguity function (BAF). While it is

often difficult to derive precise closed form expressions for the sidelobe structure, the mainlobe structure can be quite accurately modeled. The contour of the BAF's mainlobe is well approximated by a coupled ellipse, or ellipse of ambiguity (EOA). The EOA parameters determine the mainlobe width in range and Doppler as well as the degree of range-Doppler coupling. If the waveform's modulation function is known the EOA parameters may be calculated in closed form. The multi-tone sinusoidal frequency modulated (MTSFM) waveform possesses a modulation function that is represented using a Fourier series expansion. The coefficients are utilized as a set of tunable parameters that can be modified to synthesize waveforms with specific characteristics. This research presents closed form expressions for the MTSFM's BAF EOA parameters. Simulations demonstrate that the EOA parameters facilitate the design of MTSFM waveforms with desirable mainlobe and sidelobe characteristics.

11:00

5aSP8. Deconvolved conventional beamforming for a planar array and Cramer Rao bound. Yiting Zhu, T. C. Yang, Xiang Pan (School of Information and Electron. Eng., Zhejiang Univ., Zhe Rd. 38, Hangzhou, Zhejiang 310027, China), (panxiang@zju.edu.cn)

It is well known that the beam power output of conventional (delay and sum) beamforming (CBF) can be expressed as the convolution of the source distribution and the response of the beamformer to the point source. Consequently, the original source distribution can be recovered by deconvolution, yielding not only improved direction of arrival (DOA) estimation in terms of a narrower beam width and suppressed side lobes but also higher array gain than CBF (the deconvolution gain) as demonstrated by Yang for a uniformly spaced line array (ULA) [T. C. Yang, 10.1109/ JOE. 2017. 2680818]. In addition, the method is insensitive to signal mismatch and snapshot deficient problems in comparison with other high resolution methods such as minimum variance distortionless response (MVDR). In this paper, we extend the Richardson-Lucy algorithm for deconvolution to a planar array where the signal can be coming in any direction. Simulation analysis as well as experimental data will be presented to study the beam width and array gain as a function of number of receivers, the frequency of the incoming signals including narrow and broad band signals, and the ability to reject interference. We also calculate the Cramer-Rao bound to compare with the experimental result.

11:15

5aSP9. Study on the problem of coherent source resolution with same frequency for the coprime array. Jidan Mei, Dajun Sun, Yuyuan Zhou, and Junjie Shi (Acoust. Sci. and Technol. Lab.,College of Underwater Acoust. Eng., Harbin Eng. Univ., Shuisheng Bldg., Harbin Eng. University 1004, No. 145, Nantong St., Nangang District, Harbin 150001, China, meijidan@hrbeu.edu.cn)

The coprime sensor array (CSA) can achieve the same resolution capability as an equally spaced half-wave array even using fewer array elements. For single-target, CSA can achieve good anti-grating lobe effect. When there are multiple coherent sources of the same frequency, there still are overlaps between the main lobes and grating lobes of the subarray from different sources, and between the grating lobes and grating lobes of different subarray. In some cases, the strong grating lobes can interfere discriminating the azimuth and quantity of targets. This paper focuses on the multi-targets discrimination problem of CSA for the same frequency coherent source. A processing algorithm of CSA coherent sources is presented. This method uses iterative processing methods to detect and remove strong targets one by one. Then, reconstruct the spatial spectrum of each single target. Finally, recombine the spatial spectrum by each single target spatial spectrum after sidelobe suppression processing. This method can well remove the grating lobe of multiple targets. It is also applicable to the situation when the grating lobe is higher than the weak target peak under strong and weak contrast conditions. The performance of the method was simulated and the results confirmed the feasibility of the method.

11:30

5aSP10. Ultrasonic endoscope in the form of a bundle of elastic rods for imaging in aggressive liquids. Sergey Tsysar, Suren Petrosyan (Phys. Faculty, Moscow State Univ., GSP-1, 1-2 Leninskie Gory, Moscow 119991, Russian Federation, sergey@acs366.phys.msu.ru), Victor D. Svet (Acoust. Inst., Moscow, Russian Federation), and Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Modern ultrasonic imaging systems demonstrate high quality of the images obtained in non-aggressive media. Direct application of traditional systems under critical conditions, for example, at high temperature, in the presence of chemically active media or an increased level of radiation, is often impossible. We propose an approach based on the concept of an ultrasound endoscope to remotely display the area of interest through a long beam of solid rods. Each rod serves as a waveguide that carries an ultrasonic pulse from one end located in an aggressive medium to the other end located at a distance in favorable conditions. The diameter of the rods is less than half the wavelength to ensure that the ultrasonic pulses used to form the image propagate as the lowest-order longitudinal mode, which is non-dispersive and much faster than other possible modes (bending and torsional motions). The feasibility of the proposed concept is illustrated by numerical modeling and experiments. The results of studies of the 1024-channel waveguide system of megahertz range made of a bundle of stainless steel rods are presented. The obtained images of millimeter-sized scatterers in water demonstrate high resolution of the proposed system. [This work was supported by the RSF grant 17-72-10284.]

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Session 5aUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Physical Acoustics: Sediment Acoustics–Inferences from Forward Modeling, Direct, and Statistical Inversion Methods III

Charles W. Holland, Cochair
Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804

Stan E. Dosso, Cochair
School of Earth & Ocean Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada

Chair's Introduction—8:00

Invited Papers

8:05

5aUW1. Ship of opportunity noise inversions for geoacoustic parameters of a layered mud-sand seabed. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and David P. Knobles (KSA LLC, Austin, TX)

This paper considers the use of broadband noise from ship sources of opportunity in statistical inference for geoacoustic parameters of a layered seabed via a trans-dimensional Bayesian matched-field approach, with applications to data collected with a bottom-moored horizontal array in the 2017 Seabed Characterization Experiment conducted on the New England Shelf. The approach samples probabilistically over possible model parameterizations (number of seabed layers) and provides uncertainty estimates of seabed model parameters (sediment geoacoustic profiles). Approaches to accounting for (unknown) ship source level directionality when combining data from varying ship aspects to the array are addressed. Sediment parameter estimates from the inversions are compared with direct measurements from sediment cores and other geophysical data collected in the experiment area.

8:25

5aUW2. Seabed geoacoustic inversion using a hydrophone equipped underwater glider. Yong-Min Jiang (Res. Dept., NATO - STO - Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, La Spezia 19126, Italy, yong-min.jiang@cmre.nato.int), Stan E. Dosso (Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

The reflection coefficient at the sea floor as a function of grazing angle contains seabed geoacoustic properties that are crucial for predicting sound propagation accurately. NATO—STO—CMRE has developed a new technique for remote sensing the seabed reflection coefficient using underwater gliders equipped with passive acoustic payloads. This technique was applied during ONR sponsored Seabed Characterization Experiment conducted on the New England Mud Patch, USA, in March 2017. The reflected field was measured between grazing angles of 3.5 to 260, and frequency band of 2–20 kHz by an omni-directional hydrophone equipped Slocum glider, with the assistance of a broadband acoustic source. The data were analyzed both in time and frequency domains to obtain the reflection coefficient as a function of grazing angle at different frequencies. Ray tracing was also employed to fine tune the grazing angle information. Finally, Bayesian geoacoustic inversion of the reflection data at different frequencies was carried out. As a preliminary inversion attempt, the reflection data were modeled assuming a plane wave reflection coefficient. [*Work funded by NATO-Allied Command Transformation, Office of Naval Research and Office of Naval Research—Global.*]

8:45

5aUW3. Seabed roughness measurement using the multibeam-subbottom-profiler SBP120. Samuel Pinson (SHOM, SHOM, 13 rue du chatellier, Brest 29200, France, samuelpinson@yahoo.fr), Benjamin Barbier (SHOM, Brest, France), Charles W. Holland (Appl. Res. Lab., The Pennsylvania state Univ., State College, PA), and Yann Stéphan (SHOM, Brest, France)

In a recent paper [Pinson *et al*., J. Acoust. Soc. Am. 143, 2622–2631], theoretical interface echo scintillation, time-of-arrival variance, and angle-of-arrival variance have been derived as a function of the roughness autocorrelation function. Those parameters have been obtained under the Kirchhoff and small-roughness approximation by including integration over the source and receiver arrays in the Helmholtz-Kirchhoff integral. In this communication, we present the first roughness inversion results of at-sea recorded data based on the seabed-specular-echo fluctuation analysis. The data were recorded with the multibeam-subbottom-profiler SBP120 during the CALIMERO 2004 experiment campaign in the Gulf of Lion. [Research supported by the ONR Ocean Acoustics Program.]

9:05

5aUW4. Observations of resonances and antiresonances in modal propagation over muddy sediment layer. 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 (Acoust. and Mech. Eng., Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Measurements of signals generated over the New England mud patch, an area of ocean roughly 70 m deep characterized by a 10 m thick surficial mud-layer, by the Intensity Vector Autonomous Recorder (IVAR) are presented. IVAR is a bottom deployed system that measures both potential and kinetic energy roughly 1.2 m above the seafloor. Data originate from signals generated by a towed J15 source (transmission in the 50–500 Hz band), and signal underwater sound (SUS) MK-64 explosive sources, both part of the SCE17 signal set, and individual modes are identified by their unique Doppler shifts and/or through time-frequency analysis of dispersion. The density contrast between water and mud manifests a concentration kinetic energy in the water column, specifically when the water/mud interface intersects a node of the vertical mode functions. Following the work of Tolstoy (JASA 28(6), 1182–1192, 1956), data observations are fit to a representative sediment model through coupling two reduced waveguides, one representing the water column and the other the layered sediment, at their lattice points. The persistence of a resonance, which occurs at intersecting lattice points, is examined in terms of the range dependence in water depth and sediment structure.

9:25

5aUW5. Characterization of bottom parameters in the New England Mud Patch using geophone and hydrophone measurements. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), James Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin, Julien Bonnel, Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), David P. Knobles (Knobles Sci. And Anal., LLC, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

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 Woods 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 (Scholte waves). The iWaSP system consists of a source to generate the interface wave and a four-element accelerometer receive array. Modal arrivals from broadband sources (SUS and CSS) on geophones and hydrophones will be presented and compared. The individual arrivals on the sensors will be identified though seismo-acoustic modeling. Based on the propagation paths of the various waves (compressional, shear and Scholte waves), geoacoustic parameters will be estimated. Seismic data collected at the location will be used to constrain the model parameters in the inversion. Sediment data from cores, other *in-situ* measurements and inversions using other types of data will be used to compare and validate our inversions. [Work supported by Office of Naval Research.]

9:45–10:00 Break

Contributed Papers

10:00

5aUW6. Geoacoustic inversion using cross correlation function of low-frequency ambient noise in the Shallow Water 2006 Experiment. Marina Yarina, Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Abba Khouchy Ave. 199, Haifa 3498838, Israel, marina-malysheva@autorambler.ru), Jixing Qin (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Oleg A. Godin, and Tsu Wei Tan (Naval Postgrad. School, Monterey, CA)

In this paper, results of measurements and analysis of the cross correlation function (CCF) of the ambient noise field in a low frequency band (10–200 Hz) are presented. Records of the noise were obtained by single hydrophones (SHRUs) in the Shallow Water 2006 (SW06) experiment on the New Jersey continental shelf during about a month. Five SHRUs were placed approximately along a straight line perpendicular to the coastal line, distances between adjacent hydrophones were 5–8 km, ocean depth was 80–100 m. Analysis of the experimental data was carried out by calculating time-frequency diagrams of CCFs for different pairs of SHRUs, with subsequent mode filtering on the basis of the warping transform. Theoretical calculations of mode parameters for comparison with the experiment were done using different waveguide models, including the geoacoustic models previously proposed for the area of SW06 by other authors based on active source data. [Work supported by BSF-NSF grant.]

10:15

5aUW7. Passive acoustic characterization of the ocean bottom using a horizontal array of hydrophones. Tsu Wei Tan (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Monterey, CA 93943-5216, ttan1@nps.edu), Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA), Boris Katsnelson, and Marina Yarina (Marine GeoSci. Dept., Univ. of Haifa, Haifa, Israel)

Cross-correlation functions (CCF) of ambient and shipping noise recorded by two hydrophones are known to contain valuable information about the propagation environment. This paper investigates the feasibility of quantitatively characterizing the seafloor using interferometry of the ambient noise data obtained with SHARK array during the Shallow Water 2006 (SW06) experiment. SHARK array contained 32 near-bottom hydrophones, which formed a horizontal line array with nominal hydrophone separation of 15 m. Noise CCFs have been calculated between elements of the horizontal array as well as between each element of the array and a single-hydrophone near-bottom receiver approximately 3.6 km away. Sound propagation between elements of the array has been interpreted in terms of direct and reflected ray arrivals and possibly refracted (head wave) arrivals originating at interfaces within the bottom. CCFs on the longer propagation paths between the array and the single-hydrophone receiver are controlled by

waveguide effects and have been interpreted in terms of adiabatic normal modes. The warping transform of spectrograms of the measured CCFs has been used to separate the modes. Analysis and interpretation of the normal mode dispersion curves retrieved using the warping transform will be discussed. [Work supported by NSF and BSF.]

10:30

5aUW8. Geoacoustic inversion of the pressure gradient data using a single vector sensor. Junjie Shi (Acoust. Sci. and Technol. Lab., Key Lab. of Marine information Acquisition and Security, College of Underwater Acoust. Eng., Harbin Eng. Univ.; Naval Acad., Shuisheng Bldg., Nangang District, Harbin 150001, China, junejieshi@outlook.com), Dajun Sun (Acoust. Sci. and Technol. Lab., Key Lab. of Marine information Acquisition and Security, College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), Stan E. Dosso (Univ. of Victoria, Victoria, BC, Canada), and Qingyu Liu (Naval Acad., Beijing, China)

A vector sensor has the capability of obtaining the acoustic field involving the sound pressure and three-dimensional particle velocities simultaneously, which means that there exists a potential to extract deep information for further applications like direction of arrival (DOA), positioning, classification, geoacoustic inversion, etc. This paper applies a nonlinear Bayesian approach to the pressure gradient data for geoacoustic properties of South China Sea, about which the experiment took place in July of 2013 in South China Sea of nearly 90 m depth. Recorded linear frequency modulation (LFM) signals of the pressure and vertical particle velocity are windowed and processed together to estimate the pressure gradient in the vertical direction at multiple frequencies as the observation data. After the optimal sediment model is selected based on BIC (Bayesian Information Criterion) using adaptive simplex simulated annealing (ASSA), delayed rejection adaptive Metropolis (DRAM), a Markov chain Monte Carlo (MCMC) sampling method, is used not only to provide a maximum *a-posteriori* (MAP) parameters estimates but also to quantify the parameters uncertainties and inter-parameter relationships. [Work supported by the National Natural Science Foundation (Grant No. 61531012).]

10:45

5aUW9. Sequential bottom parameter estimation using blind deconvolution of sources of opportunity in ocean waveguide. Xuedong Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, zxd@mail.ioa.ac.cn), Nicholas C. Durofchalk (Georgia Inst. of Technol., Annville, PA), Lixin Wu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Karim G. Sabra (Georgia Inst. of Technol., Atlanta, GA)

This paper investigates the performance of sequential bottom parameter estimation based on ray-based blind deconvolution (RBD) of sources of opportunity using both numerical simulations and the 2016 Santa Barbara Channel (SBC) experimental recordings of shipping noise for ranges up to several kilometers. The RBD algorithm relies on estimating the unknown phase of the random sources to approximate the source-to-array channel impulse responses (CIR) by wideband beamforming along a well-resolved ray path [Sabra *et al.*, JASA, 2010, EL42-7]. The power ratio of the direct and bottom-bounced arrivals is processed to infer the bottom reflection loss and is utilized to invert for the bottom parameters. Usually, the estimated bottom reflection loss is not accurate enough as the estimated CIR is noisy. Sequential parameter estimation uses a state space model for predicting and correcting the bottom parameters as the estimated bottom reflection loss values become available. This approach is a robust estimation tool employing predictions from previous estimates and corrects stemming from models that relate bottom parameters to the bottom reflection loss. Inversions results for the SBC experiment were also performed with conventional active sources to validate the inversion obtained with RBD of sources of opportunity.

11:00

5aUW10. Segmentation of underwater acoustic images by FCM with EMD. Sakthivel M. Santhanam (Electronics and Commun., Peri Inst. of Eng. and Technol., Rajiv Gandhi Salai, OMR Kalavakkam, Chennai, Tamil Nadu 603110, India, sakthivels@ssn.edu.in) and Somasekar K. (Electronics and Commun., Peri Inst. of Eng. and Technol., Chennai, India)

In recent years, there is an expanding enthusiasm for studying the seabed structure. The investigation of seabed has numerous applications, for example, protest recognition, exploration of natural resources, angling, analyzing the composition of residues, etc. In oceanic examinations, submerged objects location is a standout amongst the most basic errands. This emphasizes the necessity of image segmentation, which divides an image into parts that have strong correlations with objects to reflect the actual information collected from the real world. Image segmentation is the most practical approach to virtually all automated image recognition systems. Clustering of numerical data forms the basis of many classification and system modelling algorithms. The purpose of clustering is to identify natural groupings of data from a large data set to produce a concise representation of a system's behaviour. In this paper, Empirical Mode Decomposition (EMD) picture upgrade procedure has been utilized to enhance the nature of sonar pictures and Fuzzy C means (FCM) clustering method is used for underwater image segmentation. The result of EMD is compared with histogram equalization and Contrast stretching. EMD with K-means and EMD with FCM is compared and the exploratory outcomes demonstrate that the proposed technique can acquire exact outcomes and enhances operational productivity.

Session 5pAA

Architectural Acoustics and Noise: Session in Memory of Murray Hodgson II

Nicola Prodi, Cochair
Dept. of Engineering, University of Ferrara, via Saragat 1, Ferrara 44122, Italy

Maureen R. Connelly, Cochair
Construction and Environment, BCIT, 3700 Willingdon Ave., Building NE03 Office 107, Vancouver, BC V5G3H2, Canada

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Invited Paper

1:15

5pAA1. Optimization of the acoustic treatments in order to achieve the Italian minimum environmental criteria (CAM) for classrooms. Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Gino Iannace, and Amelia Trematerra (Seconda Universita' di Napoli, Aversa, Italy)

A recent Italian legislation has established the Minimum Environmental Criteria (CAM) for working environments. With regard to schools, adequate acoustic comfort targets are required in terms of noise control and acoustic quality. Schools must comply with the UNI 11532 for their reverberation (T), clarity (C_{50}) and speech intelligibility (STI). In classrooms, the following values are required: T<0.7s, C_{50}>0dB and STI>0.6. In sports facilities such as gyms or swimming pools, the requirements are T<1.5 s, C_{50}>-2 dB and STI>0.5. To achieve these objectives, the insertion of acoustic treatments is often unavoidable. Acousticians commonly use the perfectly diffused theory to calculate the amount of need sound-absorbing material to comply with current legislation. The purpose of this work is to evaluate the optimal position in which to place the minimum amount of sound-absorbing treatments to reach CAM. Some university classrooms are used. These classrooms have a marble floor and plastered walls and, at 1.0 kHz, have T values between 2.5s and 4.5s, C_{50} between 3 dB and -0.5 dB, and ST between 0.34 and 0.47. Using an acoustic software, it was possible to estimate the minimum quantity and the optimal placement of the sound-absorbing panels to insert to reach the CAM of these classrooms.

Contributed Paper

1:35

5pAA2. Speech-in-noise research: From classrooms to Canadian Armed Forces operational environments. Ann Nakashima (Defence Res. and Development Canada, 1133 Sheppard Ave. West, Toronto, ON M3K 2C9, Canada, ann.nakashima@drdc-rddc.gc.ca), Jane Cai (Univ. of Toronto, Toronto, ON, Canada), and Oshin Vartanian (Defence Res. and Development Canada, Toronto, ON, Canada)

The importance of speech intelligibility in learning and occupational environments is evidenced by the abundance of research in room acoustics and auditory communication. In addition to environmental factors such as background noise and reverberation, individual factors including the presence of hearing loss, wearing of hearing aids and hearing protection devices (HPDs), and language proficiency must be considered. Previous work in these areas has provided a benchmark for the study of communication in complex, high noise environments. For Canadian Armed Forces (CAF) members, the high noise levels inside aircraft, armoured vehicles and sea vessels demand the use of HPDs and integrated radio communication systems. This presentation will summarize our research on speech understanding in noise and multi-talker environments using different communication headsets, simulated hearing loss and language proficiency. We will focus on our recent studies of communication between native (L1) and non-native (L2) English speakers, which is applicable to the CAF due to the relatively high percentage of L2 membership. L2 speakers obtained lower scores on speech-in-noise tests than L1 speakers for both radioed and face-to-face speech, despite having a high level of language proficiency. Preliminary results of an investigation of cognitive load for L2 listeners will also be presented.

Invited Papers

1:50

5pAA3. Effects of background noise, talker's voice, and speechreading on speech understanding by primary school children in simulated classroom listening situations. Mary Rudner (Dept. of Behavioural Sci. and Learning, Linköping Univ., Linköping, Sweden), Viveka Lyberg-Åhlander (Dept. of Clinical Sci., Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden), Jonas Brännström (Dept. of Clinical Sci., Logopedics, Phoniatrics and Audiol., Lund Univ., Malmö, Sweden), Jens Nirme (Lund University Cognit. Sci., Lund Univ., Lund, Sweden), Margaret K. Pichora-Fuller (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, k.pichora.fuller@utoronto.ca), and Birgitta Sahlén (Dept. of Clinical Sci., Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden)

In the primary school classroom, children are exposed to multiple factors that combine to create adverse conditions for listening to and understanding what the teacher is saying. Four experiments were conducted to investigate the combined effects of background babble noise (quiet vs. noise), voice quality (normal vs. dysphonic), and speechreading (audio-only vs. audio-visual cues) on speech understanding in 245 eight-year old children. Comprehension was tested using narratives from the test of Clinical Evaluation of Language Fundamentals. Background babble noise was composed of several children talking. Visual speech cues were presented using a digitally animated talker. Vocal loading was used to induce a dysphonic (hoarse) voice. Speech understanding was reduced by even low levels of babble noise, but compensated by visual cues. Dysphonia did not significantly reduce comprehension scores, but it was considered unpleasant. There was some evidence that performance in adverse conditions was positively associated with individual differences in cognitive executive function. Overall, these results suggest that multiple factors combine to influence speech understanding and listening effort for child listeners in the primary school classroom. The constellation of these room, talker, modality, and listener factors should be taken into account in the planning and design of educational and learning activities.

2:10

5pAA4. Research trajectories in classroom acoustics: Investigating children perception beyond accuracy. Nicola Prodi (Dept. of Eng., Univ. of Ferrara, via Saragat 1, Ferrara 44122, Italy, nicola.prodi@unife.it)

In a prospective discussion with Murray Hodgson earlier in 2003 in Ferrara, we agreed that much further insight should be gained in the perception of room acoustics for speech, and classrooms were the main area of common interest. In fact several metrics have been in use since long to assess listeners' accuracy, either objectively or subjectively. Unfortunately, these tools are not effective in tracking the listening effort, which may occur even if accuracy is high. Thus, an extension of the analysis beyond acoustically and accuracy-oriented indicators is required to grasp information on the multi-faceted construct of listening effort. In the field of audiology a framework for listening effort was developed and several psycho-physiological and behavioral quantities have been proposed. None of them is considered exhaustive, and different quantities are sensitive to peculiar aspects: for instance, the slowing down of response time in adverse conditions is thought to trace an increase of processing load. Applying this knowledge in the field of room acoustics may fill the gap in the evaluation and design of rooms for speech communication. In this work, the results achieved so far will be presented and applications in the field of classroom acoustics will be outlined too.

2:30

5pAA5. Learning and interacting in noisy classrooms: what background noise measures and subjective teacher perceptions tell us about the challenges for students who are hard of hearing. Janet R. Jamieson (Educational & Counselling Psych., & Special Education, Univ. of Br. Columbia, Faculty of Education/ECPS, University of Br. Columbia, 2125 Main Mall, Vancouver, BC V6T 1Z4, Canada, janet.jamieson@ubc.ca), Brenda Poon (School of Population and Public Health, Univ. of Br. Columbia, Vancouver, BC, Canada), and Anat Zaidman-Zait (School of Education, Tel Aviv Univ., Tel Aviv, Israel)

The negative impact of noisy classrooms can impede academic performance for even typically hearing children. The purpose of this qualitative study was to investigate the impact of noisy classrooms on the social and academic experiences of deaf and hard-of-hearing students, who are increasingly educated alongside their hearing peers. First, background noise levels were measured in 11 kindergarten-Grade 7 classrooms in which children with hearing loss were placed, during the first $1^1/_2$ hours of one school day. Based on the ongoing activities and background noise levels, predictions were made as to when the children would likely experience the most and least adverse listening conditions. Second, teachers were interviewed to obtain their perspectives of the learning and socialization experiences of the children with hearing loss. Overall, there was striking consistency between the predicted difficulties based on objective acoustic measures and perceived difficulties based on teachers' subjective perspectives. Thematic analysis of the interviews revealed the following major themes concerning the students with hearing loss: difficulty hearing instruction; missing out on social communication with peers; and difficulty recognizing and managing transitions. Overall, these findings suggest that background noise in elementary school classrooms negatively impacts listening, learning, and social interaction for students with hearing loss.

Contributed Paper

2:50

5pAA6. Verification of average absorption coefficients—A follow-up on one of Murray Hodgson's Classroom Acoustics Projects. Katrina Scherebnyj (BKL Consultants Ltd., #308,1200 Lynn Valley Rd., North Vancouver, BC V7J 2A2, Canada, scherebnyj@bkl.ca)

Twelve years ago, multivariable linear-regression analysis techniques were used with data from over a hundred UBC classrooms to estimate the "average" absorption coefficient of typical surfaces in classrooms. The purpose of the work was to obtain accurate estimates that could be used in the design of new classrooms, to better predict the early-decay time[1]. The current paper applies the results from the previous work to new classrooms, outside UBC. Detailed information is known regarding the surface types and absorption coefficients used in these classrooms. Measurements of the early decay times have also been taken. Comparisons are made between the early-decay times predicted using standard absorption coefficients, the

"average" absorption coefficients from the UBC classrooms, and measurements. [1] Hodgson M., Scherebnyj K. Technical note—*Estimation of the absorption coefficients of the surfaces of Classrooms*. Applied Acoustics 67 (2006) 936–944.

Invited Papers

3:05

5pAA7. Trajectories in classroom acoustics: The vocal behaviour of teachers. Arianna Astolfi (Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10124, Italy, arianna.astolfi@polito.it)

Classroom acoustics was one of the main research themes of Murray Hodgson, which I had the chance to know in Rome, at the ICA Conference in 2001, when I was at the beginning of my working path, and to further have as scientific converser in many other occasions. In Ferrara, in 2003, he concluded his presentation of a course on classroom acoustics with the hint that the teachers' voice problems should have been object of future studies. I have been working on this matter for seven years and this work summarises the results and the perspectives related to the assessment of teachers' vocal behaviour. In particular, teachers' voice monitoring during daily working activities has been recently based on wearable vocal analysers equipped with contact sensors, which allow for measuring parameters related to vocal effort, vocal loading and also to voice quality. Results obtained during experimental campaigns that took place in the last years, from kindergarten to university, and that involved healthy and unhealthy teachers, are presented in this work. The relationships with classroom acoustics, both in terms of noise and too low or excessive reverberations, and the subjective outcomes of the teachers, are also discussed.

3:25

5pAA8. Cross-modal effects of noise and thermal conditions on indoor environmental perceptions. Wonyoung Yang, Hyeun Jun Moon, and Myung-Jun Kim (Dept. of Architectural Eng., Dankook Univ., #318, Eng. 1 Bldg., 152 Jukjoen-ro, Suji-gu, Yongin, Gyeonggi 16890, South Korea, wyang@dankook.ac.kr)

Realistic thermal conditions with various humidity levels have been considered to examine cross-modal effects of noise and thermal conditions on indoor environmental perceptions. Subjective assessments of temperature, humidity and acoustics were conducted with 26 subjects under combined environments of seven thermal conditions (18°C: RH 30, 60%, 24°C: RH 27, 43, 65%, 30°C: RH 30, 60%), two noise types (fan and babble noises) and five noise levels (45, 50, 55, 60, and 65 dBA). Three-minute moderate noise exposure did not affect temperature or humidity sensations. However, temperature and humidity levels affected loudness, annoyance, and acoustic preferences. Men were more sensitive to hot sensations than women, and women were more sensitive to arid sensations than men. Women were more sensitive to noise levels than men. Gender differences were also found in terms of different types of noise. Men were found to be significantly less sensitive to fan noise than women. Even though psychoacoustic parameters were affected by indoor thermal conditions; thermal parameters were not affected by short-term moderate noise. The combined effect of various types of noise and temperature is still unclear, and this will be considered in a future larger cohort study.

Session 5pAB

Animal Bioacoustics: Marine Mammal Bioacoustics II

Jesse C. Turner, Chair
SMRU Consulting, 400 Emmerling Pl, Friday Harbor, WA 98250

Contributed Papers

1:00

5pAB1. Antarctic type C killer whale (*Orcinus orca*) call repertoire from McMurdo Sound, Ross Sea. Rebecca Wellard (Ctr. for Marine Sci. and Technol., Curtin Univ., GPO Box U1987, Perth, WA 6845, Australia, becwellard@gmail.com), Robert Pitman (Antarctic Ecosystem Res. Div., Southwest Fisheries Sci. Ctr., National Marine Fisheries Service, La Jolla, CA), John Durban (Marine Mammal and Turtle Div., Southwest Fisheries Sci. Ctr., National Marine Fisheries Service, La Jolla, CA), and Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA, Australia)

Killer whales (*Orcinus orca*) are found in all oceans of the world. In Antarctic waters, five ecotypes have been described, each displaying distinct differences in morphological features, foraging behaviours, habitat and diet preferences, and genetic structure. Acoustic recordings of Type C killer whales were collected between December 2012 and January 2013 in McMurdo Sound, Ross Sea, Antarctica. Spectrograms of acoustic data were examined for characteristic patterns of Type C vocalizations and a call type catalogue was produced. We measured acoustic parameters of each call type for both whistle and burst-pulse sounds and we then compared call features of Type C animals to those of other killer whales described in the Southern Hemisphere. Analysis of calls revealed that Type C killer whales produce a large number of biphonations and complex calls with multiple frequency-modulated and pulsed components. The limited accessibility of Antarctic regions year-round makes passive acoustic monitoring (PAM) a very effective tool to derive information on ecotype-specific distribution and seasonal occurrence. This study provides new information on the call repertoire of Type C killer whales, investigates the potential use of PAM to study Antarctic ecotypes, and also examines utilising call repertoire as a reliable diagnostic tool for identifying sympatric ecotypes in Antarctic waters.

1:15

5pAB2. Characteristics of Guiana dolphins (*Sotalia guianensis*) burst-pulse sounds emitted during different surface activity rates in Rio de Janeiro's coast. Mariana Barbosa (Programa de Pós Graduação em Ecologia e Evolução, Rio de Janeiro State Univ., Rua São Francisco Xavier 524, Maracanã, Rio de Janeiro, Rio de Janeiro 20550-013, Brazil, marib66@gmail.com), Lis Bittencourt, Luciana Andrade, Tatiana Bisi, José Lailson-Brito, and Alexandre d. Azevedo (MAQUA - Laboratório de mamíferos aquáticos e bioindicadores, Rio de Janeiro State Univ., Rio de Janeiro, RJ, Brazil)

Burst-pulses are still the least studied signals in delphinid acoustic repertoire. In this study, acoustic data were gathered in two Rio de Janeiro coastal bays where groups of Guiana dolphins can be found regularly. The acoustic equipment consisted of a C54XRS hydrophone (-155.8 dBV, 0.006 Hz a 203 kHz) and a Fostex digital recorder (192 kHz sampling rate). During each recording, the surface activity rate of the group was classified as being high or low. Burst-pulses were analyzed using SoundRuler software. Fifty signals were randomly selected and the values for duration (low = 143.9 $\pm$ 145.4; high = 82.2 $\pm$ 74.8), interpulse interval (low = 2.81 $\pm$ 1.92; high = 1.34 $\pm$ 1.35), number of pulses (low = 52.6 $\pm$ 55.7; high = 71.6 $\pm$ 64.5), peak frequency (low = 37.4 $\pm$ 2.5; high = 38.5 $\pm$ 3.2), and minimum frequency (low = 15.4 $\pm$ 6.5; high = 6.7 $\pm$ 5.3) were measured. Additionally, a Mann-Whitney U test compared all acoustic parameters of burst-pulses emitted during both surface activity rates. Significant differences between activity rates were found for interpulse interval ($p<0.01$) and minimum frequency ($p<0.01$). This scenario could indicate that some burst-pulse parameters are related to group arousal and behavior.

1:30

5pAB3. Seasonal patterns in marine mammal vocalizations in the western Canadian Arctic. William D. Halliday (Wildlife Conservation Society Canada, Wildlife Conservation Society, 169 Titanium Way, Whitehorse, YT Y1A 0E9, Canada, whalliday@wcs.org), Stephen J. Insley (Wildlife Conservation Society Canada, Whitehorse, Br. Columbia, Canada), and Xavier Mouy (Jasco Appl. Sci., Victoria, BC, Canada)

The Arctic marine environment is changing rapidly through a combination of sea ice loss and increased anthropogenic activity. Given that these changes can affect marine animals in a variety of ways, understanding the spatial and temporal distributions of Arctic marine animals is imperative. Here, we use passive acoustic monitoring to examine the presence of marine mammals in the western Canadian Arctic, where we have had recorders deployed near the communities of Sachs Harbour and Ulukhaktok, Northwest Territories, Canada. At both sites, we documented bowhead and beluga whales during the ice-free season, bearded seals throughout the ice-covered season and during their mating season, and ringed seals throughout the year. The sites also had different patterns in marine mammal presence, where we found whales later into the year at Ulukhaktok than at Sachs Harbour, and ringed seals vocalized much more at Ulukhaktok than at Sachs Harbour. These patterns in vocal activity at both sites help to document the presence of each species in the western Canadian Arctic, and serve as a baseline for future monitoring in the region. Next steps in this project involve deploying at more sites throughout the region to more comprehensively monitor marine mammals in the region.

1:45

5pAB4. Description of seasonal acoustic occurrences of humpback and minke whales in South Africa and Antarctica. Fannie W. Shabangu (Agriculture, Forestry and Fisheries, Fisheries Management, 13 Barlinka Ave., Tableview, Cape Town, Western Cape 7441, South Africa, fannie.shabangu@yahoo.com)

Seasonal acoustic occurrences and diel singing patterns of humpback whale (*Megaptera novaeangliae*) songs and Antarctic minke (*Balaenoptera bonaerensis*) whale are described using acoustic recordings from the west coast of South Africa and Maud Rise, Antarctica. Acoustic data were recorded from early 2014 to early 2017. Acoustic occurrences (i.e., presence) of humpback and minke whale sounds were identified through visual scrutiny of spectrograms of recorded data. Environmental conditions associated with humpback whale song occurrences were ranked according to their model-predicted relative importance. In South Africa, humpback whale songs were detected from June to December but peaked in September. In Antarctica, humpback whale songs were detected from March to May (singing peaked in April). Minke whale sounds were only recorded in 2014, between June and September in Antarctica and between September and November in South Africa. Humpback whales were more vocally active at

night in all recording sites whereas minke whales were more vocally active during the day. This is the first study to describe the seasonal acoustic occurrences of humpback and minke whales off the west coast of South Africa. Such knowledge could be essential for the conservation and management of these species in both South Africa and Antarctica.

2:00–2:15 Break

2:15

5pAB5. Development of beluga calf calls (*Delphinapterus leucas*) in the first six months of life. Audra Ames (Fundación Oceanogràfic, C/ Eduardo Primo Yúfera (Científic), 1B, Valencia, Valencia 46013, Spain, aames@oceanografic.org), Jason Wood (SMRU Consulting, Friday Harbor, WA), and Valeria Vergara (Ocean Wise Conservation Assoc., Vancouver, BC, Canada)

As part of an ongoing effort to understand beluga mother-calf communication and impacts of anthropogenic noise on neonate calls, we investigated a male beluga calf's vocal development at Oceanogràfic, using a calibrated digital hydrophone with a sampling rate of 256 kHz. His initial vocalizations were broadband pulse-trains with upper frequency limits reaching $\geq$128 kHz on his first day of life, higher than reported by earlier studies limited by lower sampling rates. Over the calf's first six months, pulsed trains were the predominant call type, increasing significantly in peak and first quartile frequencies and pulse repetition rate during this period. First and third quartile frequencies also increased significantly over the calf's first month, indicating an early and continuous shift in pulse train energy distribution towards higher frequencies. While mixed and tonal calls appeared sporadically in the calf's first month of life, they were not common until his third month, when they comprised 13% of his vocalizations. Additionally, estimated source levels of the calf's calls increased throughout his first month, but were lower than those for adult belugas. This, coupled with the progressive shift in energy distribution, has important implications regarding a neonate's ability to compensate for noise.

2:30

5pAB6. Sperm whale inter-pulse intervals and size: The anomaly called Yukusam. Jesse C. Turner, Jason Wood (SMRU Consulting, 400 Emmerling Pl, Friday Harbor, WA 98250, jt1@smruconsulting.com), and Jared Towers (Bay Cetology, Alert Bay, BC, Canada)

Yukusam is the name given to a male sperm whale who was first documented off northeastern Vancouver Island in February 2018. He spent several weeks in this area before traveling south to the inland waters of the Salish Sea in late March 2018. Sperm whale clicks have been used as a proxy to determine overall size by using the time difference of arrival between the initial noise pulse and its reflections within the spermaceti organ. Different equations have been derived in order to use this Inter-Pulse Interval (IPI) to estimate overall length. Here we use Yukusam's usual clicks recorded from the Lime Kiln hydrophone on the west side of San Juan Island (WA State, USA) to compare IPI-based length estimates with visual observations. Photo and video documentation indicate that this whale is ~15 m in length. Our initial acoustic results indicate that equations in the literature underestimate his length by at least a few meters. The potential reasons behind these anomalies are explored in this presentation.

2:45

5pAB7. Song production by the eastern North Pacific right whale, *Eubalaena japonica*. Jessica Crance, Catherine L. Berchok (AFSC/NMFS/NOAA, Marine Mammal Lab, 7600 Sand Point Way NE, Seattle, WA 98115, Jessica.Crance@noaa.gov), Dana Wright, Arial Brewer, and Daniel Woodrich (Joint Inst. for the Study of the Atmosphere and Oceans, Univ. of Washington, Seattle, WA)

This paper describes unique, stereotyped, gunshot songs produced by the eastern North Pacific right whale (NPRW, *Eubalaena japonica*). Four unique songs were documented over eight years (2009–2017) and five separate locations in the southeastern Bering Sea. Similar to songs reported for other species, each NPRW song consists of a hierarchical structure of 1-3 different repeating phrases comprised predominantly of gunshot calls (though other call types were occasionally present). Songs were detected every year from July through early January. All four song structures remained consistent over eight years. Two different songs often occurred simultaneously; the same song never occurred simultaneously at the same location. However, the detection of the same song on the same day and time at two locations 310 km apart indicates multiple animals can produce the same song. These songs were localized to individual male NPRWs using directional sonobuoys; it remains unknown if females also sing. We hypothesize that these songs may be a reproductive display similar to song in other mysticetes. Although singing has not been documented in congeneric right whale species, songs presented here fit the classification of song attributed to other cetacean species. Possible functions of these songs and their management implications will be discussed.

3:00

5pAB8. Investigation of the context of humpback whale non-song calls in the North Atlantic. Mikala Epp and Gail Davoren (Biology, Univ. of MB, 66 Chancellors Circle, Winnipeg, MB R3T 2N2, Canada, eppm34@myumanitoba.ca)

Humpback whales display behavioural plasticity in their foraging behaviours and non-song calling behaviours. Prey type appears to be associated with some of the variation in foraging behaviours and, thus, may also be related to variation in use of non-song calls related to foraging. Our goal was to compare non-song calls from two regions and identify any call types that occur in only one of the regions. We used recordings from stationary hydrophones within two foraging grounds: from the northeast Newfoundland coast, where capelin is the main prey; and the Gulf of Maine, where herring and sand lance dominate the diet. These areas are sympatric and foraging groups are thought to be from the same breeding aggregation, suggesting that call type differences would not have a genetic-basis. Most of the call types were the same across both areas, as has been found across many foraging grounds, but a small number of call types and call sequences were heard in only one of the two regions. We hope to use multi-sensor tag data and video recordings to help identify the contexts of these calls to further determine whether non-song call variation can be linked to prey type.

3:15

5pAB9. Differential hearing sensitivity in the adaptive background in a beluga whale. Vladimir Popov, Dmitry Nechaev, Alexander Supin, and Evgeniya Sysueva (Inst. of Ecology and Evolution, 33 Leninskij Prosp., Moscow 119071, Russian Federation, popov.vl.vl@gmail.com)

The ability of the auditory system to maintain high differential sensitivity in the adaptive background was investigated in a beluga whale (*Delphinapterus leucas*). Adaptive background was a train of tone pips following one another at a rate of 1 kHz. Each pip consisted of eight carrier cycles of 64 kHz. Every 128 ms, the train of pips was interrupted for 16 ms and replaced with a test signal (16-ms series of the same tone pips as in the adaptive background, but of another level). The level of the test signals varied from -15 to +20 dB relative the level of adaptive background. Evoked potentials (the rate following response, RFR) produced by the test signals were recorded. Increasing of adaptive level led to RFR thresholds growth. The 10-dB rising of adaptive signals intensity level led to 7.8 dB rising of test signal threshold. The response amplitude dependence on the test stimulus level was almost independent of the level of the adaptive background. Thus, the beluga's auditory system displayed high sensitivity to the change in acoustic signal level in the high-level background. [This study was supported by the Russian Foundation for Basic Research (Grant No. 18-04-00088).]

5p FRI. PM

Session 5pAOa

Acoustical Oceanography and Underwater Acoustics: Experimental Assessment of Theories of Sound Propagation in Sediments II

Orest Diachok, Cochair
Johns Hopkins University APL, 11100 Johns Hopkins Rd., Laurel, MD 20723

N. Ross Chapman, Cochair
School Earth and Ocean Sciences, Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada

Invited Papers

1:00

5pAOa1. Laboratory experimental assessment of models for elastic wave propagation in marine sands. Masao Kimura (Poro-Acoust. Lab., 1115-22 Miyakami, Shimizu, Shizuoka, Shizuoka 424-0911, Japan, mk45@nifty.com)

Laboratory experimental research on the models for elastic wave propagation in marine sands has been performed. The speed dispersion and attenuation of compressional and shear waves using water-saturated unconsolidated glass beads and sands have been measured. The measured compressional wave speed dispersion and attenuation could be predicted by using a gap stiffness model incorporated into the Biot model (the BIMGS model) in the range of $kd \leq 0.5$ (k: wavenumber in water, d: grain diameter) and by using the BIMGS model plus multiple scattering effects in the range of $kd \geq 0.5$. The temperature and grain size dependence of shear wave speed dispersion and attenuation has been measured. The results showed the validity of the BIMGS model. Furthermore, the empirical equations for the tortuosity, permeability, and pore radius, which are important parameters in the Biot and modified Biot models, were derived. Finally, new laboratory experiments on the dependence of the compressional wave speed and attenuation on the temperature, and the effect of grain size distribution on speed dispersion and attenuation that may shed new light on the validity of the models will be suggested.

1:20

5pAOa2. Ocean sediments and the Biot theory. Nicholas P. Chotiros (ONRG, 86 Blenheim Crescent, Ruislip HA47HB, United Kingdom, nicholas.p.chotiros.civ@mail.mil)

The surficial ocean sediment is usually a granular medium, therefore it is a porous medium and follows Biot's theory. It predicts one shear wave and two compressional waves, called the fast and slow waves. This sets it apart from the acoustic theories of uniform media (solids or fluids) which permit only one compressional wave. The fast wave is similar to the compressional wave in a solid or fluid. The slow wave is unique to porous media, it is difficult to detect and its presence in the ocean sediment is controversial. There is indirect evidence of the Biot slow wave: The deviation of the acoustic reflection coefficient of the ocean sediment from Rayleigh's reflection equation, can be explained in terms of the Biot slow wave. Direct detection of the Biot slow wave has been reported in some porous media, particularly sintered glass beads. However, its detection in unconsolidated media such as water saturated sand needs more work. Direct measurement of its properties is desirable. Simulations suggest that it could explain some unusual behavior of finely layered sediments. Finally, due to natural granularity, there should be a continuous conversion between the fast and slow waves that is missing from current models.

1:40

5pAOa3. Observation of Biot compressional waves of the second kind in granular soils. Koichi Nakagawa (Faculty of Sci., Osaka City Univ., 3-3-138 Sugimoto, Sumiyoshi-ku, Osaka City, Osaka Prefecture 558-8585, Japan, knaka@kind.ocn.ne.jp), Kenichi Soga (Dept. of Civil and Environ. Eng., Univ. of California, Berkeley, Berkeley, CA), and James K. Mitchell (Dept. of Civil and Environ. Eng., Virginia Tech, Blacksburg, VA)

The compressional wave of the second kind predicted theoretically by Biot was observed in granular soils using triaxial testing system. This theory models both the individual and coupled behavior of soil skeleton and pore fluid using the properties of pore space and elastic constants of the solid, pore fluid and solid skeleton. The wave is characterized by its large damping and hence detection of the wave in the laboratory is difficult. This difficulty was overcome by adopting highly sensitive sensor, signal stacking and appropriate filtering. The propagation of the P- and S-waves in granular materials was measured under different confining pressures and in both dry and saturated conditions. The phase of the Biot wave was the opposite to that of the P- and S- waves in all observed. According to the theory, the Biot wave results from the out-of-phase movement of the soil skeleton and pore fluid. The measured Biot wave velocities agree well with the theoretical values computed using the properties and parameters which were obtained primarily from laboratory tests. The good agreement demonstrates that the theory is valid for granular soils.

2:00–2:15 Break

Contributed Papers

2:15

5pAOa4. On an attenuation obeying a frequency power law. Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

An attenuation obeying a frequency power law scales as $|\omega|$^β, where ω is angular frequency and β is a real constant. According to the strain-hardening shear-wave equation in the grain-shearing theory, the attenuation of the shear wave supported by a marine sediment follows a frequency power law. The associated dispersion formula, which is a causal transform, predicts that the phase speed is also a power law and that the product of the phase speed and the attenuation divided by ω is a constant, independent of frequency. From the dispersion formula, along with the conditions that the phase speed and attenuation must both be positive and the propagation factor must exhibit conjugate symmetry, it follows that the exponent β can take only certain values that fall in well-defined intervals extending indefinitely with no upper or lower limit. However, to satisfy the requirement that Green's function be causal, β is further constrained to lie in the interval [0.5 1), under which condition the Green's function is maximally flat at the time the source is activated. With other values of β the Green's function is non-causal, exhibiting non-zero values at times preceding the onset of the source. [Research supported by ONR.]

2:30

5pAOa5. Viscosity-based theory of phase velocity and attenuation of sound in mud consisting of water and flocculated clay particles. Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY 12180, browne6@rpi.edu), Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., East Sandwich, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Recent theory of authors ascribes attenuation in typical marine mud sediments to be caused by viscous interaction of sea water with embedded silt particles. Influence of underlying clay matrix is regarded as passive and of minor importance. Present paper considers silt-less mud where clay particles are flocculated to a card-house structure, with the flocculation hypothesis yielding a porosity of 90%. During the passage of a sound wave, Van-der-Waals forces between platelets cause the matrix to move to-and-fro as a unit; viscous forces are insufficient to cause the matrix to move perfectly with the water. The silt-less theory assumes that the local force on the matrix is the sum of the viscous forces on the platelets in the matrix. Forces on clay particles, which are thin platelets, are given by a low-Reynolds flow theory initiated by Stokes, and further developed by Oberbeck, Lamb, and Brenner. For each particle there is a characteristic frequency inversely proportional to the platelet thickness, which turns out to be extremely high for clay. Consequently, at acoustic frequencies, the clay matrix moves nearly in lock-step with the fluid motion associated with the sound wave. The inevitable very-small slip leads to an attenuation that is proportional to the square of the frequency, but which is very small compared with that of mud with embedded silt-particles. Current idealized theory results in prediction of attenuation inversely proportional to viscosity and of nearly constant phase velocity. [Work supported by ONR.]

2:45

5pAOa6. A micro-mechanical model for the Biot theory of acoustic waves in a fully saturated granular material. Luigi La Ragione, Giuseppina Recchia (Dipartimento di Scienze dell'Ingegneria Civile e dell'Architettura, Politecnico di Bari, Via Jacini, 11, Bari, BA 70125, Italy, luigi.laragione@poliba.it), and James T. Jenkins (Dept. of Civil and Environ. Eng., Cornell Univ., Ithaca, NY)

In the context of the classical Biot theory for acoustic waves in a fully saturated granular material, we improve upon the constitutive relation of the solid phase by means of micro-mechanical modeling. This is needed to explain discrepancies on the dependency of the frequency with the sound speed attenuation and dispersion between present models and experiment. The micro-mechanical provides a more detailed description of the interaction between the particles and water and gives an expression for the stress of the entire aggregate based upon micro mechanical parameters such as coordination number (average number of contacts per particle), porosity, material properties. The aggregate is modeled as a collection of particles that are stiffer than the water; so in the particle fluid interaction, idealized by a Standard Linear Solid model, only the compressibility of the water is taken in account. The possibility to include a deviation of the particle motion from the classical affine deformation is explored. Predictions of the sound speed attenuation and dispersion are provided in the context of a uniaxial deformation.

3:00

5pAOa7. Estimating muddy seabed properties using ambient noise coherence. David R. Barclay (Oceanogr., Dalhousie Univ., Dept. of Oceanogr., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca), Dieter A. Bevans, and Michael J. Buckingham (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA)

During the Seabed Characterization Experiment, a multi-institutional field effort held at the New England mud patch, the autonomous passive acoustic lander Deep Sound made a series of ambient noise measurements from the seafloor. The instrument platform carried four hydrophones, arranged in an inverted 'T' shape with three spaced in the horizontal and two in the vertical, and landed on the seafloor with the bottom phones 30 cm above the interface. Pressure time series, vertical and horizontal noise coherence (directionality), were recorded continuously for periods of 9 hours over the acoustic bandwidth of 5 Hz to 30 kHz, along with the local temperature, conductivity, and depth. An analytical Pekeris waveguide noise model was fitted to the data in order to determine the bulk sound speed, sheer speed, density, and frequency dependent attenuation in the bottom fluid half-space. Acoustic properties of the mud were determined by comparing the data to the output of a range independent noise model, featuring a realistic multi-layered seabed. [Research supported by ONR.]

3:15

5pAOa8. Coherent processing of ocean ambient noise for sediment profiling and head-wave detection using a vertical line array in the Santa Barbara Channel. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

In this study, sediment profiling using the passive fathometer and the head-wave phenomenon are investigated. Head waves are created by waves propagating in sediments parallel to the water-sediment interface (possibly at a speed higher than the water sound speed), and propagating back in the water at critical angles. Ocean ambient noise recorded by a vertical line array is processed using a generalization of the known "passive fathometer." The technique can detect coherent time-separated arrivals at specific angles that are consistent with the propagation of head waves. The arrivals structure detected at endfire provides information on the sediment layering, while the head wave can be used to estimate the sediment sound speed, and may provide information about the attenuation. The technique is demonstrated using simulation and data from the Santa Barbara Channel Experiment of 2016, using both natural noise generated at the surface, and—for the first time—the noise generated by a ship. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

3:30

5pAOa9. MUD, the next generation in acoustic backscatter sediment measurements. Todd D. Mudge (ASL Environ. Sci. Inc, 1-6703 Rajpur Pl., Victoria, BC V8M 1Z5, Canada, tmudge@aslenv.com), Gwyn Lintern (Pacific GeoSci. Centre/NRCan, Victoria, BC, Canada), David Lemon, and Keath Borg (ASL Environ. Sci. Inc, Victoria, BC, Canada)

Detection and classification of sediments within rarely occurring turbidity flows is challenging, in part because of the acoustic absorption and scattering within the dense suspended sediments. A new calibrated acoustic backscatter echosounder, the Multifrequency Ultrasonic Device (MUD^{TM}), was deployed from 13 to 16, May 2018, at about 120 m water depth within Bute Inlet, Canada. The MUD prototype (1.2 MHz, 769 kHz and 200 kHz) was deployed in a tautline mooring which included a passive acoustic recorder, an ADCP, CT sensor and OBS sensor. The frequencies for the prototype were selected to allow for a compromise between good acoustic range and penetration into dense flows with the potential for particle size discrimination. Preliminary analysis of the data indicates the presence of three turbidity flows over a two-hour period with speeds of up to 2.5 m/s. As anticipated, the lower frequencies did better in penetrating through the dense turbidity flow. This suggests that it will be possible to use inversion of the acoustic backscatter to estimate sediment concentrations within the dense head of the turbidity flow. Multifrequency inversion techniques will be applied to the less dense portion of the flow to estimate particle size distributions.

FRIDAY AFTERNOON, 9 NOVEMBER 2018 SAANICH 1/2 (VCC), 1:00 P.M. TO 3:20 P.M.

Session 5pAOb

Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Ocean Observatories: Laboratories for Acoustical Oceanography II

Bruce Howe, Cochair
Ocean and Resources Engineering, University of Hawaii, 2540 Dole Street, Holmes Hall 402, Honolulu, HI 96822

Brendan P. Rideout, Cochair
Department of Ocean and Resources Engineering, University of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822

Invited Papers

1:00

5pAOb1. Acoustic mapping of ocean currents using moving vehicles. Chen-Fen Huang, KuangYu Chen (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenfen@ntu.edu.tw), Sheng-Wei Huang, JenHwa Guo (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., Taipei, Taiwan), and Naokazu Taniguchi (Hiroshima Univ., Taipei, Taiwan)

With the increased availability of highly maneuverable unmanned surface and underwater vehicles, abundant ocean environmental data can now be collected. Most environmental surveys by unmanned vehicles conduct point measurements of the ocean properties along survey lines. This study uses tomographic techniques to extend the survey area covered by the autonomous vehicles and obtain a synoptic ocean current distribution. An acoustic reciprocal transmission experiment was carried out on June 27, 2017, in ChaoJing Bay nearby Keelung City, where the water depth varies from 20 m to 65 m. A total of three tomographic sensors were deployed and they were installed on an AUV, a fishing boat, and a bottom-moored buoy. Reciprocal acoustic transmissions between the mobile platforms were used to estimate ocean currents, which required accounting for the Doppler effects on the acoustic arrival patterns and the resulting differential travel times. The estimated areal currents show consistency with the ADCP current measurement from the boat when the covered area was near the deeper water.

1:20

5pAOb2. Inversion of range-dependent ocean environmental parameters using radiation noises of an Autonomous Underwater Vehicle. Wen Xu, Ming Zhang, and Yuanxin Xu (Zhejiang Univ., No. 38, Zheda Rd., Hangzhou 310027, China, wxu@zju.edu.cn)

Moving platform is playing an increasingly important role in ocean observatories. We previously reported some field testing results of using the radiation noises of an Autonomous Underwater Vehicle (AUV) as the sound source to invert the range-independent SSP. To incorporate source motion effects, the forward model based on the waveguide Doppler and normal mode theory was applied to compute the replica field, and to resolve the adjacent Doppler shifted frequencies, an analytical solution of the forward model and its simplified version were obtained for arbitrary signal integration intervals with a monochromatic source. In this paper, we reformulate the matched-field inversion problem into a state-space model to track the range-dependent environmental parameters and moving source parameters along the AUV path with the constantly updated measurement equations. Performances of several sequential filters, including extended Kalman, unscented Kalman, and particle filters, are compared. Note that at each point of the track, the estimate is the average from the source to the receiver; by doing the inversion sequentially with AUV moving, one can get the parameter estimate at individual points (sections). This process is equivalent to iteration over space, while iteration over time is considered as a solution to an under-determined inverse problem.

Contributed Papers

1:40

5pAOb3. Observations and Feynman path integral analysis of acoustic fluctuations from the 2010-2011 Philippine Sea experiment. John A. Colosi, Chis Miller (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA), and Tarun K. Chandrayadula (India Inst. of Technol., Chennai, Tamil Nadu, India)

Six acoustic sources arranged in a pentagon with a central point transmitted broadband signals in the frequency range 200–300 Hz to a water column spanning vertical array in the Philippine Sea from Spring 2010 to Spring 2011. The propagation ranges were 128, 210, 224, 379, 396, and 449 km. Observations of phase and intensity variance as well as vertical coherence were obtained for several ray identified wavefronts. These observations are compared to Feynman path integral theory predictions utilizing Garrett-Munk internal wave spectral fits to *in-situ* measurements of temperature, salinity, and current. The observations and theory suggest that the three shortest ranges are in the unsaturated regime while the three longer ranges are in the partially saturated regime. Vertical coherence lengths are well predicted by theory. Implications for future experiments designed to observe internal wave spectra acoustically will be discussed.

1:55

5pAOb4. Philippine Sea deep water acoustic observations : A new test for acoustic wave propagation through random media models. Tarun K. Chandrayadula (Ocean Eng., IIT Madras, 109 B Ocean Eng., IIT Madras, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in), John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA), Peter F. Worcester, and Matthew Dzieciuch (Univ. of California San Diego, La Jolla, CA)

A primary goal of the Philippine Sea (2010–2011) experiment was to test the ability of models to predict stochastic acoustic fluctuations. The experiment had six different sources transmitting broadband pulses (200 Hz–300 Hz) to a water column spanning vertical line array. The transmission ranges varied from 130 km to 450 km, and took place across a 250 km x 250 km area. The observations are a new opportunity for two reasons. First, the source frequencies are twice that of the previous experiments that used less than 100 Hz. Second, the oceanography is more dynamic than previous sites in the North Pacific with energetic mesoscale and internal tide variability. This talk uses acoustic mode observations to analyze both broadband and narrowband statistics. Narrowband comparisons focus on mode energy estimates at different frequencies, and these results are compared with the Colosi-Morozov model. Broadband observations focus on mode pulse temporal spreads and these results are compared to predictions from a hybrid cross frequency transport theory approach.

2:10

5pAOb5. Information content of ship noise on a drifting volumetric array for passive environmental sensing. Jacquelyn S. Kubicko (U.S. Coast Guard, U.S. Coast Guard Acad., New London, CT 06320, jkubicko@gmail.com), Christopher M. Verlinden (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Karim G. Sabra (Georgia Inst. of Technol., Atlanta, GA), Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Brendan Nichols, James S. Martin (Georgia Inst. of Technol., Atlanta, GA), and Aileen Fagan (U.S. Coast Guard, New London, CT)

This study investigates the information content of ship noise received on a drifting volumetric array of hydrophones in shallow water marine environments for the purposes of conducting acoustic thermometry or other environmental inversions. Passive inversions for physical oceanographic parameters are conducted using travel time differences, determined by cross-correlating ship noise received on hydrophones suspended beneath drifting buoys. Ships are tracked using the Automatic Identification System (AIS). Information content gained from the inversion is assessed using traditional a-posteriori error analysis. Numerical simulations using a standard normal mode propagation model are used to test limitations of the proposed approach with respect to frequency band, drifting receiver configuration, precision and accuracy of the inversion results, along with sensitivity to environmental and position mismatch. Performance predictions using this model are compared with results from a field experiment using at-sea data collected off the coast of New London, CT in Long Island Sound. Information gathered using passive acoustic inversion methods on drifting arrays can be used to constrain general circulation models (GCMs), in coastal environments, where ship noise is ubiquitous, environmental data are sparse, and the oceanography is dynamic and important for understanding large-scale ocean processes.

Invited Paper

2:25

5pAOb6. Using ships of opportunity for array element localization and relative channel impulse response estimation. Kay L. Gemba, Jit Sarkar, Bruce Cornuelle, William S. Hodgkiss, and William A. Kuperman (MPL/SIO, UCSD, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92093-0238, gemba@ucsd.edu)

The uncertainty of estimating relative channel impulse responses (CIRs) obtained using the radiated signature from a ship of opportunity is investigated. The ship observations were taken during a 1.4 km (11 min) transect during the Noise Correlation 2009 (NC09) experiment. Beamforming on the angle associated with the direct ray-path yields an estimate of the ship signature, subsequently used as a matched filter. Relative CIRs are estimated every 2.5 s independently at three vertical line arrays (VLAs) for a total of 270 observations per VLA. The estimated relative arrival-time uncertainty is inversely proportional to source bandwidth and CIR signal-to-noise ratio, and reached a minimum standard deviation of 5 μs (approximately 1 cm). The direct-path relative arrival-times are used to construct time series for each VLA element across the 11 min observation interval. The overall structure of these time series compares favorably with that predicted from an array element localization (AEL) model that exhibits sensitivity on the order of centimeters. The short-term standard deviations calculated on direct-path (7 μs) and bottom-reflected-path (17 μs) time series are in agreement with the estimated arrival-time accuracies. The implication of these observed arrival-time accuracies in the context of making sound speed perturbation and bottom-depth estimates is discussed.

Contributed Paper

2:45

5pAOb7. Acoustics on the roadmap for an Integrated Arctic Observing System. Hanne Sagen and Stein Sandven (Polar Acoust. and Oceanogr. Group, Nansen Environ. and Remote Sensing Ctr., Thormøhlensgt 47, Bergen 5006, Norway, hanne.sagen@nersc.no)

The current status of the ocean *in situ* observing system in the Arctic are addressed in the EU funded Integrated Arctic Observation System (INTAROS). We describe how multipurpose acoustic systems can contribute to fill gaps and to establish an optimized Pan Arctic Ocean Observing System. A brief presentation of the steps from a series of experiments in the Eastern Arctic towards the upcoming pan-arctic Coordinated Arctic Acoustic Thermometry Experiment are given. How we can use this experience to ensure that multipurpose Arctic observation system will get on in roadmap for an Arctic Observing System? Any sustainable ocean observing systems in the Arctic depend on long-term funding, and that funding mechanisms other than research programs should be used for this. To achieve this, we need to engage with stakeholder groups outside the scientific community. How can this be done?

Invited Paper

3:00

5pAOb8. The feasibility of a multipurpose acoustic network in Baffin Bay. Eric Rehm (Takuvik, Université Laval, 1045, av. de la Médecine, local 2078, Pavillon Alexandre-Vachon, PQ, QC G1V 0A7\6, Canada, eric.rehm@takuvik.ulaval.ca), Brian D. Dushaw (Nansen Environ. and Remote Sensing Ctr., Seattle, Washington), Lee E. Freitag (Woods Hole Oceanographic Institution, Woods Hole, MA), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA), Scott Carr (JASCO Appl. Sci., Dartmouth, NS, Canada), Thomas Dakin (Ocean Networks Canada, Victoria, BC, Canada), David Fissel (ASL Environ. Services, Inc., Victoria, BC, Canada), and Garry J. Heard (Defense R&D Canada, Dartmouth, NS, Canada)

Operating autonomous underwater vehicles at high latitudes is a challenge because ice cover prevents the use of GPS or data communications. As a result, our scientific observations are biased towards late spring, summer, and early autumn when ships can navigate and autonomous platforms can safely surface. To address this problem, we studied the feasibility of a basin-scale multipurpose acoustic network called the "Baffin Bay Acoustic Navigation and Communication System" (BBANC). BBANC would deploy broadband low frequency sources and receivers, offering one-way communication, acoustic positioning, and acoustic thermometry services. Passive acoustic listening elements would support the study of marine mammal communication and ambient noise from ships, ocean-based resource exploitation, and ice dynamics, as well as gate acoustic source operation in the presence of marine mammals. We describe the challenges and design parameters for such a system, as well as define additional acoustic and remote sensing measurements required to complete a system design. Drawing from a large database of Baffin Bay hydrography, we present simulations of under-ice sound speed conditions, ice properties derived from satellite remote sensing and upward looking sonar data, and modelled acoustic propagation paths in an ice-covered Baffin Bay. We also assess the feasibility of non-coherent and coherent communication.

FRIDAY AFTERNOON, 9 NOVEMBER 2018 COLWOOD 1/2 (VCC), 1:00 P.M. TO 3:45 P.M.

Session 5pPA

Physical Acoustics: General Topics in Physical Acoustics II

Anthony L. Bonomo, Chair

Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817

Contributed Papers

1:00

5pPA1. Strong nonlinear coupling between two cavitation bubbles in an acoustic field. Vikash Pandey (The Res. Ctr. for Arctic Petroleum Exploration (ARCEx), UiT The Arctic Univ. of Norway at Tromsø, Postboks 1080, Blindern, Oslo 0316, Norway, vikashp@ifi.uio.no)

Strong nonlinear coupling comes into play between cavitation bubbles as a result of their individual oscillatory behavior in a strong acoustic field. Such a nonlinearity may play a significant role in the evolution of a bubble; from its inception to the violent collapse, in particular, in a system of multi-bubbles. The nonlinearity may also drive the bubble system to a chaotic regime, hence making the system inherently unpredictable, though deterministic. Ironically, nonlinearity has often been ignored in most of the scientific studies due to the complexity that it introduces in the theory and the resulting numerical solution. The nonlinear coupling in the simplest case of two cavitation bubbles is studied using the Keller-Miksis equation (KME). The governing KME is solved numerically assuming spherical symmetry and coupling of the bubble oscillations. Also, the role of initial conditions is examined in sufficient detail to explore the additional aspects of bubble dynamics. Further, it is found that the secondary Bjerknes force differs significantly from the predictions when nonlinearity is ignored. It is believed that these results may have implications in industries where the phenomena of acoustic cavitation, bubble cloud dynamics, and sonoluminescence are encountered.

1:15

5pPA2. Void fraction inference in cavitating fuel injector flows. Lucas K. Watson, Huy K. Do, Sheryl Grace, Emily Ryan, and R. G. Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, Lkwatson@bu.edu)

It has been shown that hydrodynamic cavitation within fuel injectors plays a significant role in their performance, with the desirable effect of broadening the resultant fuel spray. Experiments are challenging owing to the relatively small geometries, high pressure, and high Reynold's number (Re) associated with such flows. Previous studies have observed cavitation in optically transparent nozzles at slower flows. By utilizing acoustic and vibration measurement techniques cavitation activity may be measured in a steel fuel injector at more practical Re flows used in applications. We report here experimental measurements taken using a laser vibrometer and a commercial fuel injector. Previous studies have demonstrated a resonant frequency shift as a function of injection pressure. Among competing hypotheses, our working hypothesis is that this shift is the result of mass unloading of cantilever mode oscillations of the fuel injector. The dynamic void fraction caused by cavitation activity within the fuel injector can then be inferred from the measured frequency shift. We report measurements of mode shapes and frequencies for static and flowing fuel injectors as functions of the flow rate.

1:30

5pPA3. Simulated cavitation noise from strong nonlinear coupling in a multi-bubble system. Vikash Pandey (Ctr. for Ecological and Evolutionary Synthesis (CEES), Dept. of BioSci., Univ. of Oslo, Postboks 1080, Blindern, Oslo 0316, Norway, vikashp@ifi.uio.no)

The mutual nonlinear coupling between cavitation bubbles in a bubble cloud has often been blamed for the complex nature of the cavitation noise. This noise is undesirable for two reasons. First, it affects the source air-gun signature which is crucial in later stages of seismic signal processing. Second, the generated noise may fall in the same frequency regime which is used by some types of whales for communication, and therefore interfere. The complete nonlinear theory of bubble interaction is used to solve the governing Keller-Miksis equation (KME) to simulate the experimentally observed cavitation noise. A qualitative comparison is made with the results from other scientific investigations and interesting inferences are drawn. It is believed that the findings reported here may have implications in the air-gun array design in seismic exploration industry since air-guns are the indirect source of cavitation noise. It may also have an impact on the study of communication systems in marine mammals.

1:45

5pPA4. Study of the refraction of sound by shear and boundary layers. David Francoz (AeroDynam. Aeroelasticity Acoust. Dept., ONERA, The French Aerosp. Lab, ONERA, 29 Ave. de la Div. Leclerc, Châtillon 92320, France, david.francoz@onera.fr), Renaud Davy, Patrice Malbequi (AeroDynam. Aeroelasticity Acoust. Dept., ONERA, The French Aerosp. Lab, Châtillon, France), Emmanuel Julliard (Turbomachinery Noise Sources Team Acoust. Dept., Airbus Operations S.A.S., Toulouse, France), and Michel Roger (Ctr. for Acoust. Res., ECL, CNRS-LMFA, Écully, France)

Performing accurate acoustic measurements in aerodynamic wind tunnel is a challenge of great interest for the aeroacoustics community. Hence aeroacoustic measurements could be obtained in the same facility under the same aerodynamic conditions. This paper shows the refraction effects to be taken into account during aeroacoustics measurements in a closed test section, including flows up to high subsonic Mach numbers. In closed wind tunnel, the wall-mounted microphones are either flush-mounted or recessed in a cavity to avoid perturbations from the boundary layer. Besides, due to the wall, interference pattern generated by reflections occur. Thus, the acoustical wave front is strongly modified compared to the open test section configuration. Numerical computations, using the ONERA's sAbrinA-v0 CAA code solving the Euler's equations, allow us to accurately characterize the influence of a shear layer (recessed microphones) and of a boundary layer (flush-mounted microphones) and measurement setups, on the radiation from a monopolar source in the presence of a rigid wall. The CAA computations are made in the range 1 kHz to 10 kHz and for different boundary/shear layers thicknesses. The paper details a comparison between the Amiet's model and the CAA computations. The comparisons are in a good agreement despite the absence of a wall in the Amiet's model.

2:00

5pPA5. Guided shear waves and solitons in nonlinear viscoelastic materials. Emilien F. Dilly (Joint Dept. of Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 45, Rue d'Ulm, Paris, Ile de France 75005, France, efa.dilly@gmail.com), François Coulouvrat (Institut Jean Le Rond d'Alembert, CNRS, Paris, France), Bharat Tripathi, and Gianmarco Pinton (Joint Dept. of Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC)

The nonlinear propagation of elastic waves in soft solids, such as gelatin or brain, can easily give rise to shocks. Previous experimental and theoretical studies have described planar, focused, one dimensional, two dimensional polarized shear wave propagation in these media, as well as acoustic non linear propagation in wave guides. Here we present the behavior of guided shear waves in solid plates. A model based on a quasi-modal wave decomposition is described as well as a Fourier-based numerical solution that takes into account nonlinear, attenuating, dispersive, guided propagation. A mixed conventional time-space and Fourier domain numerical method is used to determine harmonic generation with propagation. These numerical solutions are validated by a comparison with experimental data for plane waves in gelatin and brain. It is then shown how the nonlinearity and dispersion due to the guiding geometry can act together to assist the formation of shocks. The model also predicts, similarly to fiber optics physics, that nonlinearity can give rise to a self phase modulation, which under specific conditions can be compensated by the dispersion resulting from the waveguide. It is shown that the model therefore predicts the existence of solitons -both theoretically and numerically.

2:15–2:30 Break

2:30

5pPA6. Reflection of finite amplitude acoustic wave from a vapor-liquid interface. Takeru Yano (Osaka Univ., 2-1, Yamada-oka, Suita 565-0871, Japan, yano@mech.eng.osaka-u.ac.jp)

We numerically investigate the reflection of finite amplitude acoustic wave from a vapor-liquid interface and resulting evaporation or condensation flow induced by the reflected wave on the basis of the kinetic theory of gases. As an initial condition, we consider the finite amplitude plane acoustic wave propagating in a vapor bounded by the liquid layer of the same molecule as the vapor. The governing equation of wave motion in the vapor is the Boltzmann-Krook-Welander equation and the boundary condition at the interface is the complete condensation condition. The Boltzmann-Krook-Welander equation is numerically solved with a finite difference method. As a result, we clarify the reflection law of finite amplitude wave at the interface, which determines an evaporation or condensation flow established after the wave reflection.

2:45

5pPA7. Acoustic streaming in a channel a moderate streaming Reynolds number. Charles Thompson, Kavitha Chandra (ECE, UMASS Lowell, 1 Univ Ave., Lowell, MA 01854, charles_thompson@uml.edu), and Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., East Sandwich, MA)

In this work, the generation of acoustic streaming in a rigid walled channel is examined. At low values of the streaming Reynolds number, the time-averaged fluid motion in the channel follows that given by Rayleigh. However, departure from the aforementioned result ensues as the magnitude of the streaming Reynolds number increases. Higher order nonlinear corrections to the Rayleigh streaming solution is given and are expressed in terms of a regular perturbation sequence in nondimensional particle amplitude. It is shown that the reduction in the amplitude of the axially directed streaming velocity is a function of the streaming Reynolds number.

3:00

5pPA8. Formation of cylindrical shock wave in two-phase liquid layer with free surface. Valeriy Kedrinskiy and Ekaterina Bolshakova (Physical HydroDynam., Lavrentyev Inst. of HydroDynam., Russian Acad. of Sci., Lavrentyev Prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru)

Numerical model for formation of 1D cylindrical shock-wave (SW) in a liquid layer with a free surface is considered. The liquid states are pure water and distilled water containing free micro-bubbles (1.5 ö¥m, 10^6 cm^{-3}). The SW initiation is performed on the axis by giving the pulse of mass velocity as the exponent for maximal amplitudes from 60 to 20 m/s. A two-phase mathematical model is the system of equation describing average pressure, velocity and density (including the Rayleigh-type equation). For pure water, the distributions of maximal amplitudes both from the axis along the radius for SW and from free surface up to the axis of symmetry for rarefaction wave (RW) were calculated. The distribution of maximal amplitudes of positive SW and negative RW along the radius appears to be completely symmetric. It was shown, the SW amplitude decreases proportionally to $r^{-0.45}$ (within 3 cm from axis) and then asymptotics ($r^{-0.72}$) is registered. The increase of RW amplitude during propagation to the axis is a cumulative effect. In two-phase model of distilled water the cavitation begins behind the RW front. Beginning from free surface, the volumetric gas concentration increases in 300 initial values (maximum velocity 60 m/s).

3:15

5pPA9. Thickness measurement of rigid porous material through reflected acoustic waves at Darcy's regime. Mustapha Sadouki (Departement des Sci. de la Matière, Université Djilali Bounaama à khemis-miliana, Université Djilali Bounaama à Khemis-Miliana, Rte. Thenia el Had, Ain Defla, Khemis-miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz)

In this work, an inverse method is proposed for measuring the thickness of air-saturated of rigid porous material using the reflected acoustic waves at Darcy's regime. The equivalent fluid model is considered. The interactions between the structure and the fluid are taken into account in two frequency response factors; the dynamic tortuosity of the medium introduced by Johnson *et al.* and the dynamic compressibility of the air introduced by Allard. A simplified expression of the reflection coefficient is obtained at the Darcy's regime (very low frequencies), this expression is independent of the frequency and depends only on the thickness and the flow resistivity of the porous medium. The inverse problem is solved numerically by minimizing between the simulated and experimental reflected signals; the reconstructed values found of the thickness of porous samples with different resistivity are in good agreement to those obtained using direct measurements.

3:30

5pPA10. Acoustic wave propagation in orthotropic porous silicon with specific pore shapes. Xiaoyue Gong, Julien Bustillo, Laurianne Blanc, and Gaël Gautier (GREMAN UMR 7347, INSA CVL, Univ. of Tours, 3 Rue de la Chocolaterie, INSA-CVL, Blois, Loire-et-Cher 41000, France, xiaoyue.gong@insa-cvl.fr)

In the literature, porous materials are normally treated as isotropic media. In this case, analytical solutions for wave velocities and attenuations as a function of frequency have been given by M. A. Biot. Nevertheless, porous silicon (PSi) synthesized using electrochemical manufacturing is generally highly anisotropic. Then, it can be considered as orthotropic. In order to optimize acoustic applications using PSi, it is necessary to take account to this anisotropy. In early works, by using FEM simulations, skeleton elastic constants have been computed for five PSi pore shapes and different porosities. In this work, the "Gedanken experiments" have been implemented and compliance tensors have been calculated. By inverting the compliance tensor, the elastic parameters of fluid-saturated PSi have been calculated. Moreover, geometrical parameters such as tortuosity, permeability and characteristic length of pores have been simulated. The wave equations can be deduced by combining Biot's theory and Christoffel's equations, and have been solved by using the simulation results. Then, the phase velocity and attenuation variations, according to frequency, were studied. The results for five different pore shapes have been compared and the differences show that the shape strongly influences wave propagation.

FRIDAY AFTERNOON, 9 NOVEMBER 2018 RATTENBURY A/B (FE), 1:00 P.M. TO 3:30 P.M.

Session 5pSP

Signal Processing in Acoustics: General Topics in Signal Processing III

Dieter A. Bevans, Chair
Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Dr., La Jolla, CA 92093-0238

Contributed Papers

1:00

5pSP1. Ultrasonic phased array total focusing method based on sparse deconvolution. Bei Yu, Yujian Mei (Zhejiang Univ., No. 38, Zheda Rd., Xihu District, Hangzhou City, Zhejiang Province 310027, China, yubei@zju.edu.cn), Haoran Jin (Nanyang Technolog. Univ., Hangzhou, ZheJiang, China), Eryong Wu, and Keji Yang (Zhejiang Univ., Hangzhou City, China)

Ultrasonic phased array total focusing method (TFM) has been widely used in the field of non-destructive testing (NDT) as an advances post-processing imaging algorithm. It shows great potential in defect detection and characterization. However, due to the small number of excitation elements and large amount of collecting data, its temporal resolution and imaging efficiency need to be improved. To overcome these disadvantages, an ultrasonic phased array total focusing method based on sparse deconvolution has been proposed in this paper. Based on the actual situation of ultrasonic testing, a regularized sparse deconvolution model has been established. The Bayesian theory is used to add low-probability prior information to the deconvolution model. At the same time, the sparse deconvolution optimization algorithm proposed in this paper is used to solve the sparse deconvolution model. Compared with ordinary phased array total focusing method, this algorithm reduced the calculation amount and improved the temporal resolution. The performance of the proposed method is proved by simulation and experimental comparison. The results verify that the proposed method has higher resolution and imaging efficiency.

1:15

5pSP2. High-resolution space-time spectral estimation method for array signal based on deconvolution. Wei Guo and Shengchun Piao (Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, guowei1030@hrbeu.edu.cn)

Conventional beamforming and classical spectral estimation methods are widely used in array signal processing to estimate the angular and frequency spectrum, respectively. But the spectra provided by these algorithms appear as low resolutions and high side lobes because of the limited array aperture and the number of snapshots. Then a multi-dimensional window function is defined, which is composed of the snapshot number and array aperture. Based on the theories of space-time joint spectral estimation, the multi-dimensional power spectrum about the azimuthal angle and frequency of array signal can be deduced as the convolution of the multi-dimensional power spectrum of the true signal and the power spectrum of the multi-dimensional window function. Therefore, the deconvolution algorithm can be introduced to remove the influence from the window function to recover the power spectrum of the true signal which is considered to be received from the infinite time and space. It is illustrated from simulations and experimental data processing that the deconvolution method can greatly improve the angular and frequency resolutions of the multi-dimensional spectrum, suppress the side lobes and provide extra signal-to-noise ratio, which can be applied for multi-target identification and weak signal detection.

1:30

5pSP3. Time delay estimation based on cross power spectrum with Doppler compensation. Xue Han (Harbin Eng. Univ., Rm. 1317 Shuisheng Bldg., No.145 Nantong St., Harbin 150001, China), Wei Guo (Harbin Eng. Univ., Harbin, Heilongjiang, China), and Shengchun Piao (Harbin Eng. Univ., Harbin, China, psc828@foxmail.com)

Time delay estimation is an important topic in underwater acoustic field. However, the Doppler phenomenon caused by relative movement between the source and receivers can arise errors in time delay estimation. In order to deal with this problem, a time delay estimation method based on cross power spectrum with Doppler compensation in frequency domain is proposed in this paper. The proposed method separates the Doppler compensation from the time delay estimation rather than estimates them jointly. The results of simulations and experimental data processing show that this method can improve the estimation accuracy of time delay with the influence of Doppler phenomenon.

1:45

5pSP4. Study on inversion of medium velocity change distribution based on code interference. zhiguang qi (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., Xi'an 710072, China, 18073150314@163.com), Hong Hou (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., Xi'an, Shaanxi, China), and Nansha Gao (School of Marine Sci. and Technol., Northwestern PolyTech. Univ., Xi'an, China)

We introduce a method that was proposed in recent years to monitoring micro geological change. The contents of this technology include the extraction of green function from background noise, the theory of code wave interference and the distribution of sensitive nuclei. We extract the green's function from the background noise and get time delay, and then establish the sensitive nuclear distribution according to the physical characteristics of the measured medium and get the medium velocity variation distribution finally. Now we have carried out the code wave interference experiment of Panjin oilfield .In the process of oil exploitation, the vibration wave velocity can be changed due to the change of medium, which can be monitored by the code wave interference. 10 seismographs are set in the range of 1km x 3km oilfield and continuous collection of data is not less than two months, and map of the speed of the oil field working face as a function of time is obtained through the background noise code wave interference method.

2:00

5pSP5. Simulated target insertion for automatic target recognition performance estimation. Jeannine Abiva and Julia Gazagnaire (Naval Surface Warfare Ctr., Panama City Div., 110 Vernon Ave., Panama City, FL 32407, jeannine.abiva@navy.mil)

To support automatic target recognition (ATR) algorithm development as well as predict their performance in various operational environments, there is a need for more realistic sonar data that captures environmental effects that are not only difficult to model using physical acoustical models but expensive to measure in the real world. Previous target insertion efforts have focused on high frequency imaging sonar. As such, these efforts effectively combine only the magnitudes of the real and simulated data. This effort will focus on low frequency sonar, and the goal is to retain the phase by combining the complex data where possible. Maintaining the phase allows us to take advantage of the resonant frequency information available in the lower frequency bands, which is lost when the magnitudes are summed. In this effort, several approaches to inserting simulated target data into real sonar data will be investigated. Typically, there are no shadows in the low frequency images; therefore, none of the approaches investigated here take shadows into account. For each method, an analysis will be carried out to evaluate how the simulated targets behave compared to real targets. These analyses will be performed on real and simulated image snippets, gradients of snippets, and the frequency content of the target scattering response or acoustic color plots. The analysis will include several statistical measures which will be used to compare snippets of real and simulated targets as well as backgrounds of their immediate surroundings.

2:15–2:30 Break

2:30

5pSP6. Assessing the accuracy of head related transfer functions in a virtual reality environment. Joseph A. Esce and Eoin A. King (Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117) (eoking@hartford.edu)

While graphics and visual representations for Virtual Reality (VR) systems are very well developed, the manner in which audio signals and acoustic environments are recreated in a VR system is not. In the case of audio spatialization, the current approach makes use of a library of standard head related transfer functions (HRTFs), i.e., a user selects a generic HRTF from a library, with limited personal information. It is essentially a "best-guess" representation of that individual's HRTF. This limits the accuracy of audio developments for virtual reality. This paper reports on results from localization tests to determine the capabilities of a generic HRTF used in a VR environment. Volunteers entered a VR world, and an invisible sound source made a short bursts of white noise at various position in the room. Volunteers were asked to point to the location of the sound source, and results were captured to the nearest millimeter using the VR's motion tracking system. It is proposed that future versions of this experimental methodology will enable the development of a pseudo-personalized HRTF, unique to each individual VR user.

2:45

5pSP7. Estimating the flight parameters of a moving airborne source using one hydrophone. Boxuan Zhang, Yixin Yang, Xijing Guo, and Ningning Liang (Northwestern PolyTech. Univ. School of Marine Sci. and Technol., 127 West Youyi Rd., Beilin District, Xi'an 710072, China, kiwinan@163.com)

The low frequency tone noise of the turboprop aircrafts can be discovered more easily under water than the broadband continuum spectrum noise due to its higher intensity. Ferguson and Lo provided a method based on a single hydrophone for estimating the four flight parameters, including the source frequency, velocity, altitude and CPA time (the time when the source passes the closet point of approach), using the variation of the instantaneous frequency (IF) of a single tone over time [Ferguson and Lo, IEEE J. Oceanic Eng., 24(4):424–435 (1999)]. Therefore, the performance of their method is largely determined by the precision of the IF estimates, which were obtained by using the short-time Fourier transform (STFT). In this paper, the polynomial chirplet transform (PCT), which models the change of the IF over time as a polynomial, is used instead of STFT to provide more accurate IF estimates. Moreover, multiple tones are used to improve the precision of the flight parameter estimates. The performance improvement is demonstrated by simulations and an experiment carried out near the Sanya coast in May, 2018.

3:00

5pSP8. Source localization based on matrix filter and sparse asymptotic minimum variance. Yahao Zhang, Yixin Yang, and Long Yang (Northwestern PolyTech. Univ. School of Marine Sci. and Technol., 127 West Youyi Rd., Beilin District, Xi'an 710072, China, yahaoZhang@126.com)

Wideband direction of arrival (DOA) estimation plays an important role in passive sonar signal processing. Recently, sparsity-based DOA estimation method has attracted considerable attention because of its high resolution in the condition of few snapshots and low signal-to-noise ratio. However, the localization accuracy is seriously affected by the interferences. Matrix filter (MF) has been widely used in passive sonar systems as a useful tool to passband the targets-of-interest while attenuating the interferences, but the output of the MF seriously affected subsequent DOA estimation when the power of the interferences after filtering is still stronger than the weak targets. In this paper, a method based on MF and sparse asymptotic minimum variance (SAMV) is given to localize the weak targets in a strong interference environment. The given method improves the ability of SAMV on weak targets localization and achieves high localization accuracy even in the condition that the power of the interferences after filtering remains stronger than the weak targets, which is verified by simulation and experimental results.

3:15

5pSP9. Bottom-laid targets imaging and classification using a synthetic circular array. Ting Zhang and T. C. Yang (Zhejiang Univ., Office 431, Xindian Bldg., Yuquan Campus, 38 Zheda Rd., Xihu district, Hangzhou 310027, China, zhang_ting@zju.edu.cn)

Detection and classification of bottom-laid targets in littoral area represents a serious problem for the navies of the world. It has been shown based on past acoustic research that the backscattered field carries a lot of information about the subject in terms of frequency and angular distribution if ensonified over a wide range of distance and can thus be used to different a mine from a stone or one type of mine from the other, referred to as acoustic color image. A circular array is more advantageous compared to a planar array as circular patterns maximize the aperture, and due to its high angular resolution, one can search and classify multiple targets at the same time. While it is difficult to deploy a large circular array since one does not know where is the target and the circular array needs to have a large diameter. Therefore, in this work, one will demonstrate effective bottom-laid targets imaging and classification using a synthetic circular array, which is created by a towed line array. We will study the concept of the proposed system and make simplified assumptions in order to obtain a crude estimate of the system performance.

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Austin, TX
Email: austinacousticalsociety@gmail.com

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Brigham Young Univ.
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Camilo Perez
Univ. of Washington
Seattle, WA 98105
Email: campiri@uw.edu

CHICAGO
Shane Kanter
Threshold Acoustics LLC
Chicago, IL 60604
Email: skanter@thresholdacoustics.com

UNIVERSITY OF CINCINNATI STUDENT CHAPTER
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Univ. of Cincinnati
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Email: richkt@mail.uc.edu

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Columbia College Chicago
Chicago, IL 60605
Email: lronsse@colum.edu

FLORIDA
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Florida State Univ.
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Email: richard.morris@cci.fsu.edu

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Georgia Institute of Technology
Atlanta, GA 30332-0405
Email: acousticalsocietygt@gmail.com

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Reuter Associates, LLC
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Alexander S. Douglass
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MUSIC CITY
Scott Hawley
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NARRAGANSETT
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Email: dbacoustics@cox.net

UNIVERSITY OF NEBRASKA STUDENT CHAPTER
Jonathan Weber
Univ. of Nebraska
Omaha, NE 68182-0681
Email: Jonryanweber@gmail.com

NORTH CAROLINA
Noral Stewart
Stewart Acoustical Consultants
Rayleigh, NC
Email: noral@sacnc.com

NORTH TEXAS
Peter F. Assmann
Univ. of Texas-Dallas
Richardson, TX 75083
Email: assmann@utdallas.edu

NORTHEASTERN UNIVERSITY STUDENT CHAPTER
Zach Neveu
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Jordan Vasko
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Erica Hoffman
Email: hoffme2@rpi.edu

SAINT LOUIS
Mike Biffignani
Email: mjbsk8@msn.com

UPPER MIDWEST
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David Braslau Associates, Inc.
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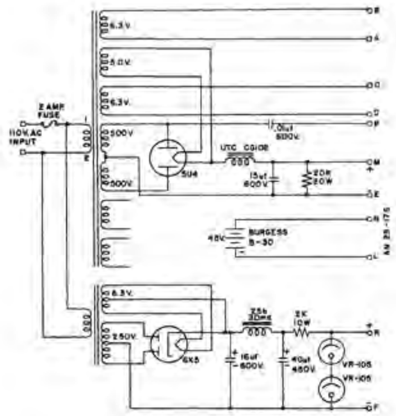

The Harvard Acoustics Research Laboratory was established in 1946 to support basic research in acoustics. Research results were disseminated formally by means of reports called technical memoranda (TMs). This CD includes the 61 reports issued between 1946 and 1971, when the contract with the Office of Naval Research was completed.

About half the TMs are doctoral theses in report form though some incorporate substantial additions. Most of the other half represent output by the postdoctoral fellows. The collection is introduced by David T. Blackstock of the University of Texas at Austin and brief bios for all of the TM authors are included.

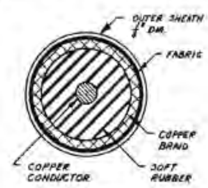

Fig.5. Schematic section showing cable makeup.

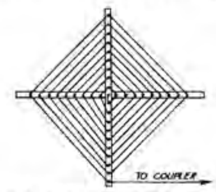

Fig.6. Configuration of the spiral mounting for the cable.

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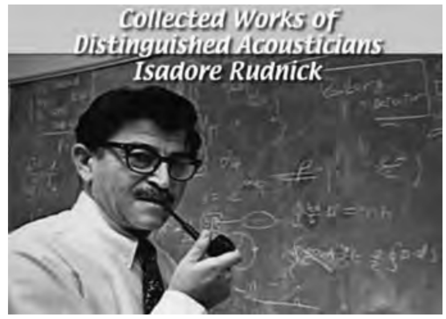
Collected Works of Distinguished Acousticians

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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